Networking your world

VoIP ATA NVP-300 SERIES

USER'S MANUAL

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Maximum signal rate derived from IEEE Standard specifications. Actual data throughput will vary. Network conditions and environmental factors, including volume of network traffic, building materials and construction, and network overhead lower actual data throughput rate.

Safety Warnings

For your safety, be sure to read and follow all warning notices and instructions before using the device.

- **DO NOT** open the device or unit. Opening or removing covers can expose you to dangerous high voltage points or other risks. ONLY qualified service personnel can service the device. Please contact your vendor for further information.
- **Use ONLY** the dedicated power supply for your device. Connect the power cord or power adapter to the right supply voltage (110V AC in North America or 230V AC in Europe).
- **DO NOT** use the device if the power supply is damaged as it might cause electrocution. If the power supply is damaged, remove it from the power outlet. DO NOT attempt to repair the power supply. Contact your local vendor to order a new power supply.
- **Place** connecting cables carefully so that no one will step on them or stumble over them. DO NOT allow anything to rest on the power cord and do not locate the product where anyone can work on the power cord.
- **DO NOT** install nor use your device during a thunderstorm. There may be a remote risk of electric shock from lightning.
- **DO NOT** expose your device to dampness, dust or corrosive liquids.
- **DO NOT** use this product near water, for example, in a wet basement or near a swimming pool.
- **Connect ONLY** suitable accessories to the device. Make sure to connect the cables to the correct ports.
- **DO NOT** obstruct the device ventilation slots, as insufficient airflow may harm your device.
- **DO NOT** place items on the device.
- **DO NOT** use the device outside, and make sure all the connections are indoors. There may be a remote risk of electric shock from lightning.
- **Be careful** when unplugging the power, because the transformer may be very hot.
- **Keep** the device and all its parts and accessories out of children's reach.
- **Clean** the device using a soft and dry cloth rather than liquid or atomizers. Power off the equipment before cleansing it.
- This product is **recyclable**. Dispose of it properly.

Foreword

NVP-300 series allows you to make telephone calls using a computer network, over a data network like the Internet. NVP-300 series converts the voice signal from your telephone into a digital signal that travels over the internet then converts it back at the other end so you can speak to anyone with a regular phone number. When placing a VoIP call using a phone with an adapter, you'll hear a dial tone and dial just as you always have. Some gateways at the recipient side can convert phonecall back into voice signal, you need go through a VOIP to PBX gateway if you'd like the phone call from VOIP to regular phone number.

VOIP Solution

Internet Voice, also known as Voice over Internet Protocol (VoIP), is a technology that allows you to make telephone calls using a broadband Internet connection instead of a regular (or analog) phone line. Some services using VoIP may only allow you to call other people using the same service, but others may allow you to call anyone who has a telephone number - including local, long distance, mobile, and international numbers. Also, while some services only work over your computer or a special VoIP phone, other services allow you to use a traditional phone through an adaptor.

Basic Services

What is Proxy Servers?

Proxy servers handle two functions:

1. Accept registrations from the SIP user agents,
2. Proxy requests and responses between user agents.

Registration is the process by which a user agent tells the proxy who it is and at what IP address and port that it can be reached via SIP. Registration usually expires within a finite period (e.g., 60s or 3600s) and the UA shall renew their registration periodically before the last registration expires. When a user agent initiates a call, it sends a SIP INVITE request to the proxy server and indicates the target recipient of the call. The proxy server then consults a database to determine where to forward the request to the destination user agent. The proxy server can request authentication credentials from the user agent before granting the service.

The credentials are computed by the user agent based on a pre-provisioned password and a challenge "nonce" dynamically generated by the proxy server per request. This mechanism prevents unauthorized user agents from getting IP Telephony services through the proxy server. SIP proxy servers are operated by the IP Telephony service provider and resides at the service provider's domain. They may be implemented in many different ways. They can be stateless, stateful, or B2BUA. Stateless proxies do not maintain states of each call; they simply proxy the requests and responses between the user agents. Hence they are the simplest, most scalable, but provide the least types of services. Advanced IP Telephony services are possible with these proxies only with intelligent user agent devices that are capable of delivering these services without proxy intervention. Stateful proxies maintain the call state of each call and can provide more intelligent services at the expense of more processing load per call. B2BUA proxies process every request and response from the user agents and are capable of providing very advance services even with relatively simple user agent devices. Obviously B2BUA proxies have the highest processing load per call.

What is SIP Services?

Today's PSTN offers a large number of enhanced services in addition to basic phone services. Most of the services offered by the PSTN are accessed by the subscribers through their telephone sets. The subscribers provide their input by talking into the handset, pressing the keypad, the switch hook or flash button, while the PSTN presents instructions/information/confirmation to the subscribers through a variety of audio tones, beeps and/or announcements. The PHONE ADAPTER supports a comparable range of services via a similar user interface in order to make the IP Telephony service transparent to subscribers.

The PHONE ADAPTER is fully programmable and can be custom provisioned to emulate just about any traditional telephony service available today. This ability to transparently deliver legacy services over an IP network coupled with the availability of Internet connected devices (PCs. PDA, etc.) and browsers opens up a new world of potential offerings that a provider can use to differentiate their service and grow their business.

Enhanced Services

Enhanced Services are provided in addition to Basic calling services and accessed by way of a touchtone phone through a series of menus. Since the service enabled by the NVP-300 series are Internet in nature, these enhanced services can be made better by offering users a web browser based interface to control certain aspects of some or all services.

Caller ID

In between ringing bursts, the NVP-300 series can generate a Caller ID signal to the attached phone when the phone is on-hook. Calling Line Identification Presentation (CLIP) Some subscribers will elect to always block their Caller ID information, yet there may be a circumstance where sending Caller ID information for a particular call is desired, i.e. trying to reach a party that does not accept Caller ID blocked calls. The subscriber activates this service to send his Caller ID when making an outgoing call. To activate the service, the subscriber enters the corresponding * or # code prior to making the call. This service is in effect only for the duration of the current call. Calling Line Identification Restriction (CLIR) – Caller ID Blocking.
The subscriber activates this service to hide his Caller ID when making an outgoing call. To activate the service, the subscriber

enters the corresponding * or # code prior to making the call. This service is in effect only for the duration of the current call.

Call Waiting

The subscriber can accept a call from a 3rd party while engaging in an active call. The NVP-300 series shall alert the subscriber for the 2nd incoming call by playing a call waiting tone.

Disable or Cancel Call Waiting By setting the corresponding configuration parameter on the NVP-300 series, the PHONE ADAPTER supports disabling of call waiting permanently or on a per call basis.

Call-Waiting with Caller ID in between call waiting tone bursts, the NVP-300 series can generate a Caller-ID signal to the attached phone when it is off hook.

Call Transfer

Three parties are involved in Call Transfer: The transferor, transferee, and transfer target. There are 2 flavors of call transfer: Attended Transfer (Transfer with consultation) and Unattended Transfer ("Blind" Transfer).

Attendant Transfer the transferor dials the number of the transfer target, then he hangs up (or enters some * or # code) when the transfer target answers or rings to complete the transfer.

Unattended or "Blind" Transfer
The transferor enters some * or # code and then dials the number of the transfer target to complete the transfer (without waiting for the target to ring or answer).

Call Hold

Call Hold lets you put a caller on hold for an unlimited period of time. It is especially useful on phones without the hold button. Unlike a hold button, this feature provides access to a dial tone while the call is being held.

Three-Way Calling

The subscriber can originate a call to a 3rd party while engaging in an active call.

Three-Way Ad-Hoc Conference Calling

The NVP-300 series can host a 3-way conference and perform 3-way audio mixing.

Call Forwarding

These services forward all the incoming calls to a static or dynamically configured destination number based on three different settings. These services may be offered by the NVP-300 series or by the SIP proxy server. They can be activated by entering certain * or # code, followed by entering a telephone number to forward calls to. The NVP-300 series provides audio instructions to prompt the user for a forwarding number and confirms that the requested service has been activated.

Speed Dial Phone

The NVP-300 series supports user programming of up to ten phone number, local,international or emergency numbers and/or IP addresses for fast and easy access.

IVR (Interactive Voice Response)

Provides mechanism for information to be stored and retrieved using voice and a touchtone telephone.

Auto provisioning

Provisioning servers are used to provision the subscriber user agent devices,e.g. the NVP-300 series. When a subscriber signs up for IP Telephony service, he selects an appropriate service level and enters his personal information including billing information. This information is processed by the provisioning server and stored into the service provider's customer database. The provisioning server generates a device profile based on the subscriber's choice of options. The device profile, which is list of configuration parameters, is downloaded into the NVP-300 series from the provisioning server. The NVP-300 series can be configured to contact the provisioning server periodically to check for any update of the device profile, which may include a firmware

upgrade or configuration modification to the NVP-300 series.(For detailed information please reference to P.57)

Auto Answer(NVP-300SO/SSO only)

Auto Answer will auto change from FXS to FXO(PBX) when call in without reply over setting time. Call in persons can call you again through FXO when NVP-300 series auto change to FXO and hearing a dial tone.

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Chapter 1. Unpacking Information

1.1 Check List

Carefully unpack the package and check its contents against the checklist.

Package Contents

- VOIP ATA 2FXS/FXS+FXO/2FXS+FXO with Router Adapter
- 1 x User's Manual CD
- 1 x AC to DC **12V** Power Adapter

Note1:
Please inform your dealer immediately for any missing or damaged parts.
If possible, retain the carton including the original packing materials.
Use them to repack the unit in case there is a need to return for repair.

Note2:
Do not use sub-standard power supply. Before connecting the power supply to the device, be sure to check compliance with specifications. The NVP-300 series of the power supply at least use DC12V/1A.

Chapter 2. Installation

2.1 Hardware Installation

This chapter describes how to install the NVP-300 series and establishes network connections. You may install the NVP-300 series on any level surface (e.g, a table or shelf). However, please take note of the following minimum site requirements before you begin.

2.2 Pre-installation Requirements

Before you start actual hardware installation, make sure you can provide the right operating environment,including power requirements, sufficient physical space, and proximity to other network devices that are to be connected. Verify the following installation requirement:

- Power requirements: DC12V/1A or above.
- The NVP-300 series should be located in a cool dry place, with at least 10cm/4in of space at the front and back for ventilation.
- Place the NVP-300 series out of direct sunlight, and away from heat sources or areas with a high amount of electromagnetic interference.
- Check if network cables and connectors needed for installation are available

2.3 General Rules

Before making any connections to the NVP-300 series,note the following rules:

- **Ethernet Port (RJ-45)**

  All network connections to the Modem Ethernet port must be made using Category 5 UTP for 100Mbps;

  Category 3,4 UTP for 10Mbps

  No more than 100 meters of cabling may be use between the MUX or HUB and an end node.

- **Phone Port (RJ-11)**

  All Phone set connections to the RJ-11 Port made using 24~26 Gauge phone wiring.

2.4 Connecting the NVP-300 series

The NVP-300 series has two ETHERNET port which support connection to Ethernet operation. The devices attached to these ports must support auto-negotiation or 10Base-T OR 100Base-TX unless they will always operate at half duplex.

Using WAN port connect to devices such as XDSL modem or router. Using LAN connect to devise such as NIC or switch.

The NVP-300 series has two RJ-11 ports, which support connection to two of analog phone set.Using Phone 1or Phone 2 port to connect to analog phone set.

1. The NVP-300 series RJ-11 ports support 2 of traditional analog phone set with IP voice transmissions. Please see (Figure 1.1~1.3).
2. The NVP-300 series RJ-45 ports support 10/100Mbps auto negotiation and auto MDIX functions, one WAN Port for connecting to XDSL Modem or Cable modem, one LAN port for connecting to PC networking Card or switched/HUB. Please see (Figure 1.1~1.3).

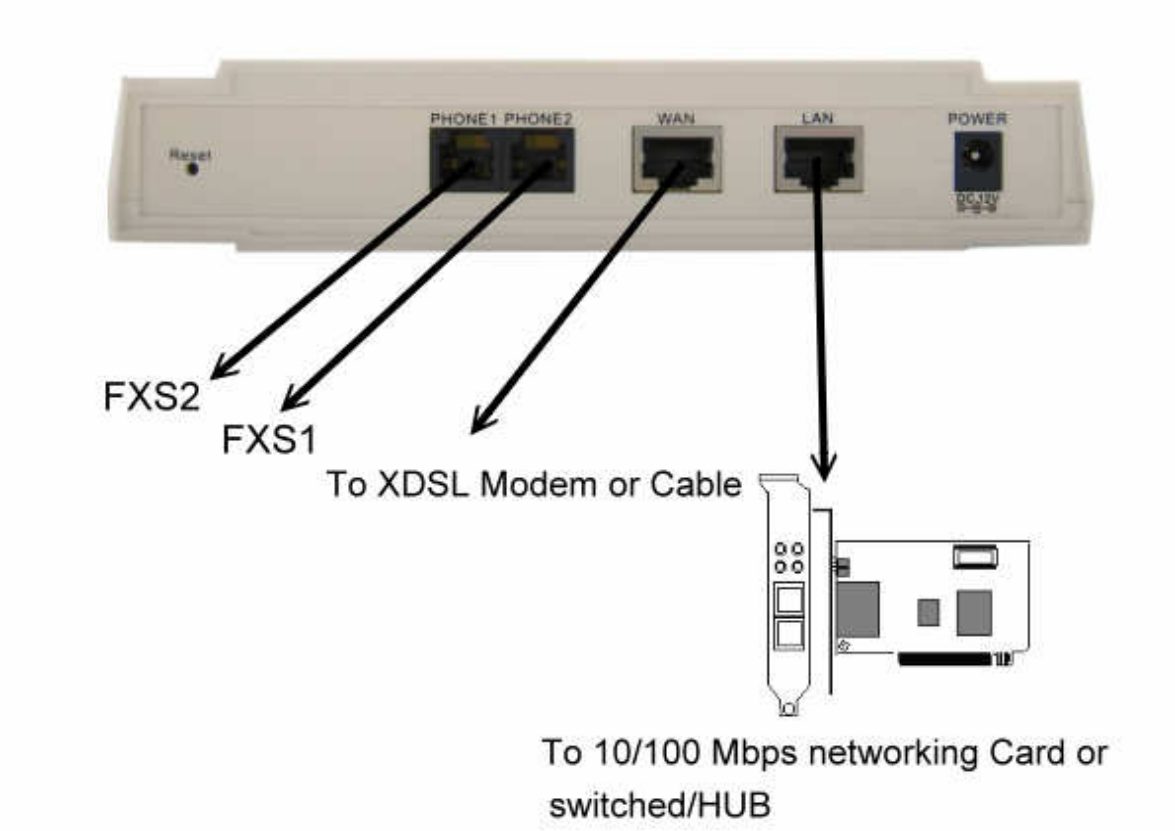


Figure1.1 NVP-300SS use as adapter to connect RJ-11 and the LAN card inside the PC

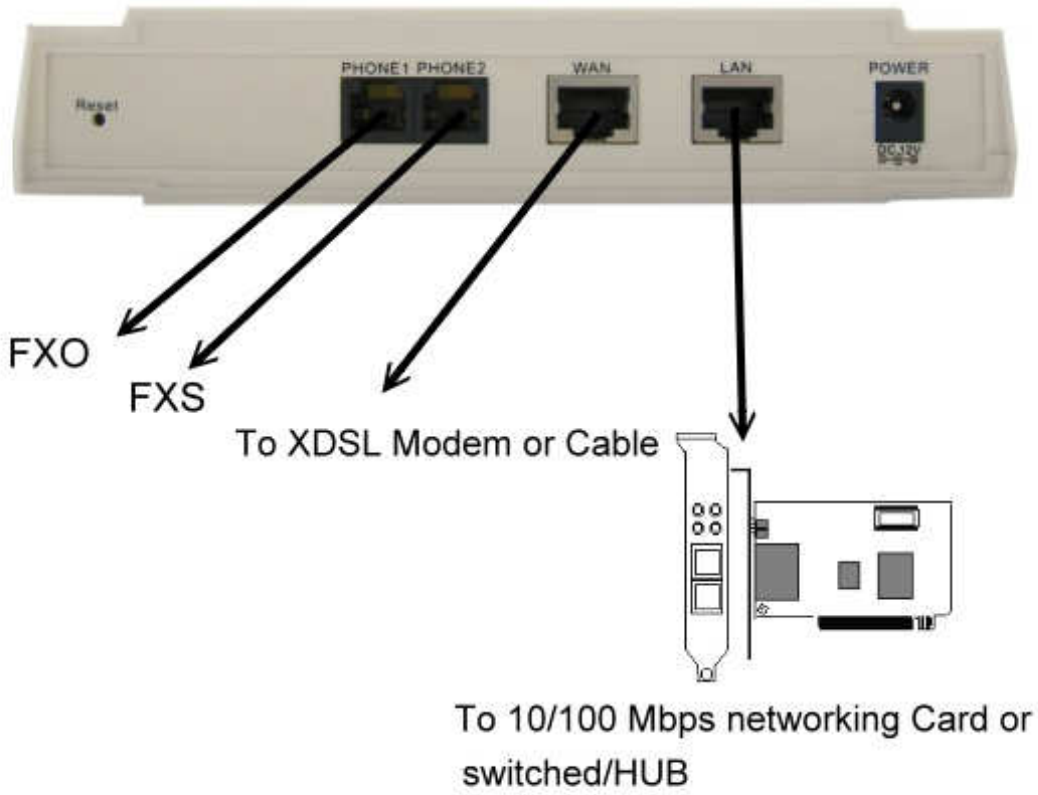


Figure1.2 NVP-300SO use as adapter to connect RJ-11 and the LAN card inside the PC

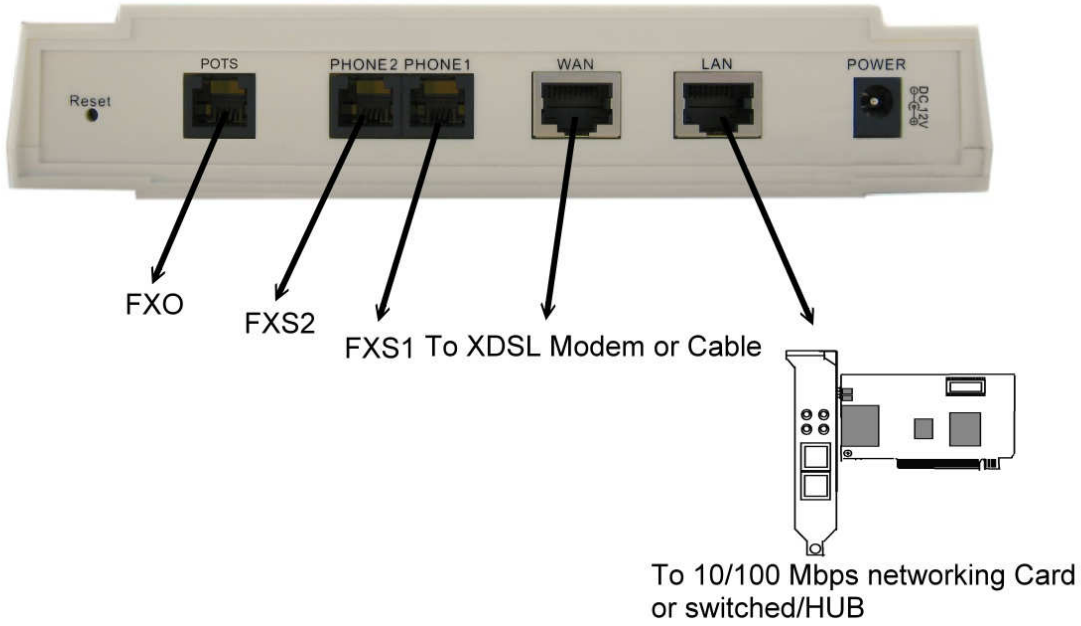


Figure1.3 NVP-300SSO use as adapter to connect RJ-11 and the LAN card inside the PC

3.Do not plug a RJ-11 phone jack connector into the Ethernet port (RJ-45 port). This may damage the modem Instead; use only twisted-pair cables with RJ-45 connectors that conform to FCC standard.

Chapter 3. Hardware Description

This section describes the important parts of the NVP-300 series. It features the front indicators and rear connectors.

3.1 Front Panel

The following figure shows the front panel.

Figure Chapter 2.1 NVP-300SS

Figure Chapter 2.2 NVP-300SO

Figure Chapter 2.3 NVP-300SSO

3.2 Front Indicators

The NVP-300SS has **Five** LED indicators. The following Table shows the description. (Table 2-1)
At a quick glance of the front panel, it will be easy to tell if the modem has power, signal from its Ethernet RJ-45 port or there is phone line signal RJ-11port

Table 2-1 LED Indicators Description and Operation

LEDs	Color	Status	Descriptions
PWR Power Good LED	Green	Steady	It will light up (ON) to show that the product power is good, and system reset OK.
Ethernet (WAN/LAN LED)	Green	Steady	Each RJ45 station port on the Ethernet is assigned a LED light for monitoring port "Good Linkage". LED is normally OFF after the power on operation, but will light up steadily to show good linkage. And Flashing to show data transmission.
		Blinking (LINK/ACT)	
Phone set dialing status (P1/P2 LED)	Green	Steady	RJ11 station port on the PHONE1/PHONE2 is assigned a LED light for dialing function OK, when you pick up handset If LED flashing indicate getting a ringing.
		Blinking(Ringing)	

The NVP-300SO has **Six** LED indicators. The following Table shows the description. (Table 2-2)

Table 2-2 LED Indicators Description and Operation

LEDs	Color	Status	Descriptions
PWR Power Good LED	Green	Steady	It will light up (ON) to show that the product power is good, and system reset OK.
RDY LED	Green	Steady	It will light up when SIP server register OK.
Ethernet (WAN/LAN LED)	Green	Steady	Each RJ45 station port on the Ethernet is assigned a LED light for monitoring port "Good Linkage". LED is normally OFF after the power on operation, but will light up steadily to show good linkage. And Flashing to show data transmission.
		Blinking (LINK/ACT)	
Phone set dialing status (PHONE LED)	Green	Steady	RJ11 station port on the PHONE is assigned a LED light for dialing function OK, when you pick up handset If LED flashing indicate getting a ringing.
		Blinking (Ringing)	
POTS(FXO)	Green	Steady	For connecting to the PBX

The NVP-300SSO has **Six** LED indicators. The following Table shows the description. (Table 2-3)

Table 2-3 LED Indicators Description and Operation

LEDs	Color	Status	Descriptions
PWR Power Good LED	Green	Steady	It will light up (ON) to show that the product power is good, and system reset OK.
Ethernet (WAN/LAN LED)	Green	Steady	Each RJ45 station port on the Ethernet is assigned a LED light for monitoring port "Good Linkage". LED is normally OFF after the power on operation, but will light up steadily to show good linkage. And Flashing to show data transmission.
		Blinking (LINK/ACT)	
Phone set dialing status (P1/P2 LED)	Green	Steady	RJ11 station port on the P1/P2 is assigned a LED light for dialing function OK, when you pick up handset If LED flashing indicate getting a ringing.
		Blinking(Ringing)	
POTS(FXO)	Green	Steady	For connecting to the PBX

3.3 Rear Panel

The following figure shows the rear connectors

Figure Chapter 2.5 NVP-300 SS (SO) Rear Connectors

Figure Chapter 2.6 NVP-3000SS0 Rear Connectors

NVP-300 series Rear Side Connectors

Table 2-4 Rear Side Connectors

Connectors	Description	Type
Rest button	For reboot system	push switch
POTS	For connecting to the PBX	RJ-11(NVP-300SO/SSO only)
Phone1/ Phone 2	For connecting to the telephone or Fax	RJ-11
WAN	For connecting to XDSL/Cable modem	RJ-45
LAN	For connecting to a PC networking card or switched/HUB	RJ-45
Power	For connecting to DC12V/1A or above power adapter	2.0m/m plug

Note:

Power On:

1. Check if the modem is properly connected
2. Verify the power LED is steadily on

Chapter 4. Introduction

This user's manual is for NVP-300 series VoIP terminal adapter. This user's manual will explain the IVR instruction, web configuration and command line configuration for the NVP-300 series. Before using the NVP-300 series, some setup processes are required to make the NVP-300 series work properly. Please refer to the Setup Menu for further information.

4.1 Hardware Overview

The NVP-300 series has the following interfaces for Networking, telephone interface, LED indication, and power connector.

- Two RJ-45 Networking interface, these two interfaces support 10/100Mps Fast Ethernet.you can connect one RJ-45 Fast Ethernet port to the ADSL or Switch, and connect the other one to your computer.
- Two RJ-11 Type analog telephone jack interfaces. You can connect two analog telephones terminal adapter.
- LED Indication: There are three LED indicators in the NVP-300 series to show the Power, Register, and Off-Hook indication.

4.2 Software Overview

Network Protocol	Tone
• SIP v1 (RFC2543), v2(RFC3261) • IP/TCP/UDP/RTP/RTCP • IP/ICMP/ARP/RARP/SNTP • TFTP Client/DHCP Client/ PPPoE Client • Telnet/HTTP Server • DNS Client • NAT/DHCP Server	• Ring Tone • Ring Back Tone • Dial Tone • Busy Tone • Programming Tone
Codec	**Phone Function**
• G.711: 64k bit/s (PCM) • G.723.1: 6.3k / 5.3k bit/s • G.726: 16k / 24k / 32k / 40k bit/s (ADPCM) • G.729A: 8k bit/s (CS-ACELP) • G.729B: adds VAD & CNG to G.729	• Volume Adjustment • Speed dial key • Phone book • Flash
Voice Quality	**IP Assignment**
• VAD: Voice activity detection • CNG: Comfortable noise generator • LEC: Line echo canceller • Packet Loss Compensation • Adaptive Jitter Buffer	• Static IP • DHCP • PPPoE
	Security
	• HTTP 1.1 basic/digest authentication for Web setup • MD5 for SIP authentication (RFC2069/ RFC 2617)
	QoS
Call Function	• ToS field
• Call Hold	**NAT Traversal**

• Call Waiting • Call Forward • Caller ID • 3-way conference	• STUN
	Configuration
DTMF Function	• Web Browser • Telnet • IVR/Keypad
• In-Band DTMF • Out-of Band DTMF • SIP Info	**Firmware Upgrade**
SIP Server	
• Registrar Server (three SIP account) • Outbound Proxy	• TFTP • Console • HTTP

Chapter 5. IVR Interface for the NVP-300 series

You can use the PSTN phone to configure the NVP-300 series. Please follow the instruction to configure your terminal adapter.

Group	IVR Action	IVR Menu Choice	Parameter(s)	Notes
Function	FXS to FXO	**0***	None	NVP-300 series FXS+FXO/2FXS+FXO only. Press hook switch once when you would like to return FXS
Setting	Set DHCP client	**#111#**	None	The system will change the WAN port to DHCP Client type
Setting	Set Static IP Address	**#112xxx*xxx*xxx*xxx#**	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	WAN port DHCP Client will be disabled and WAN port will change to the Static IP type. Set WAN port IP Address
Setting	Set Network Mask	**#113xxx*xxx*xxx*xxx#**	Enter value using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	Must set Static IP first. Set WAN port Network Mask
Setting	Set Gateway IP Address	**#114xxx*xxx*xxx*xxx#**	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	Must set Static IP first. Set WAN port Gateway IP Address
Setting	Set Primary DNS Server	**#115xxx*xxx*xxx*xxx#**	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	Must set Static IP first. Set WAN port Primary DNS Server IP Address
Info	Check IP Address	**#120#**	None	IVR will report the LAN port IP address
Info	Check IP Type	**#121#**	None	IVR will report the WAN Port DHCP is enabled or disabled.
Info	Check the Phone Number	**#122#**	None	IVR will report current in use VoIP number

Info	Check Network Mask	**#123#**	None	IVR will report the WAN Port network mask
Info	Check Gateway IP Address	**#124#**	None	IVR will report the WAN Port gateway IP address
Group	**IVR Action**	**IVR Menu Choice**	**Parameter(s)**	**Notes**
Info	Check Primary DNS Server Setting	**#125#**	None	IVR will report the WAN Port Primary DNS server IP Address.
Info	Check IP Address	**#126#**	None	IVR will report the WAN port IP address
Info	Check Firmware Version	**#128#**	None	IVR will report the firmware version
Setting	Set Codec	**#130+[1-8]#**	1:G.711 u-Law, 2: G.711 a-Law, 3: G.723.1, 4: G.729a, 5: G.726 16K, 6: G.726 24K, 7: G.726 32K, 8: G.726 40K,	You can set the codec you want to the first priority.
Setting	Set Handset Gain	**#131+[00~15]#**	Handset Gain from 0~15	You can set the Handset gain to proper value, default is 10
Setting	Set Handset Volume	**#132+[00~12]#**	Handset Volume from 0~12	You can set the Handset volume to proper value, default is 10
Setting	TFTP Server IP Address	**#135#**	Set Auto config TFTP Server IP Address	You can set the TFTP Server IP address
Setting	FTP Server IP Address	**#136#**	Set Auto config FTP Server IP Address	You can set the FTP Server IP address
Setting	Auto config mode	**#137+[0~2]#**	0: Disable, 1: TFTP mode, 2: FTP mode	You can set the Auto configuration mode, 0: Disable, 1: use TFTP Server, 2: user FTP Server
Function	enable call waiting	**#138#**	None	Enable Call waiting
Function	disable call waiting	**#139#**	None	Disable call waiting
Function	unlock keypad	**#190#**	None	you have to unlock keypad first, then you

				can change the setting by keypad.
Function	lock keypad	**#191#**	None	lock keypad.
Setting	IP mode	**#192#**	Set defualt use IP mode	Only support 1S1P, provide setting change default setting to IP mode
Setting	PSTN mode	**#193#**	Set defualt use PSTN mode	Only support 1S1P, provide setting change default setting to PSTN mode
Function	Reboot	**#195#**	None	The system will reboot automatically.

Group	IVR Action	IVR Menu Choice	Parameter(s)	Notes
Function	Factory Reset	**#198#**	None	System will automatically Reboot and restore to default setting. WARNING: ALL "User-Changeable" NONDEFAULT SETTINGS WILL BE LOST! This will include network and service provider data.

Chapter 6. Setup the NVP-300 series by Web Browser

The NVP-300 series provides a built-in web server. You can use Web browser to configure the NVP-300 series. First please input the IP address in the Web page. In the end of IP address, please add the port number ":9999". Ex:http://192.168.123.1:9999

6.1 Login.

◆ Please input the username and password into the blank field. The default setting is:

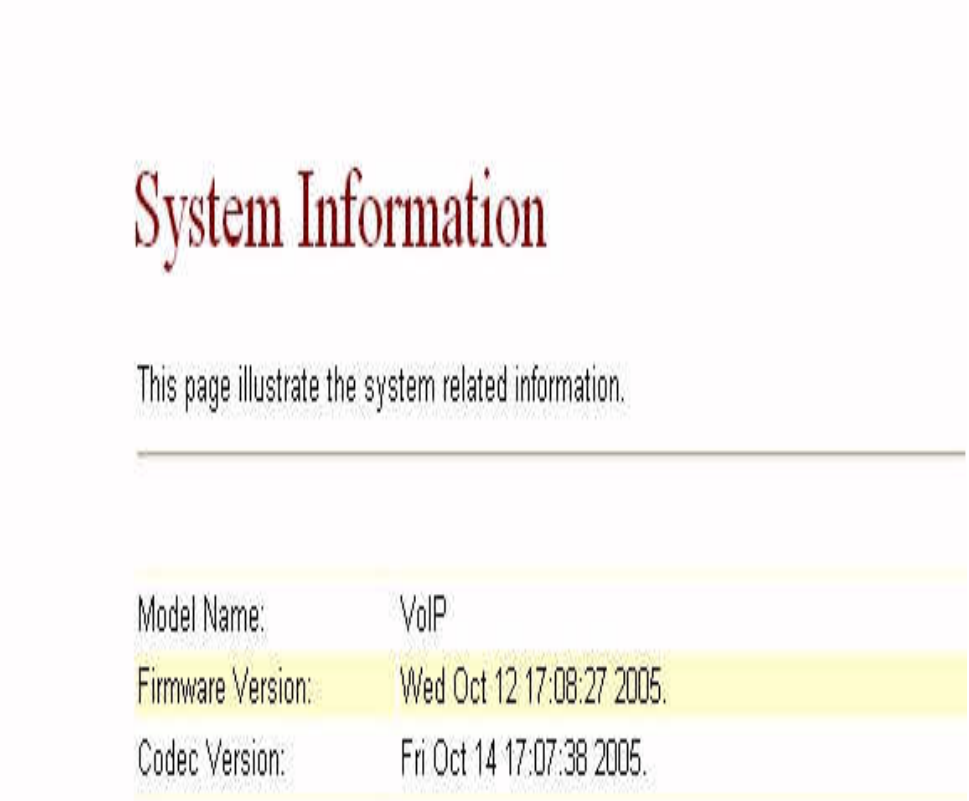
1.For Administrator, the username is: root; and the password is: test. If you use the account login, you can configure all the setting.
2.For normal user, the username is: user; and the password is: test. If you use the account login, but you can not configure the SIP setting.

◆ Click the "Login" button will move into the NVP-300 series web based management information page.

◆ If you change the setting in the Web Management interface, please do remember to click the "Submit" button in that page. After you finished the change of the setting, click the "Save" function in the left side, and click the Save Button. When you finished the setting, please click the Reboot function in the left side, and click the Reboot button in that page. After the system restart, all the setting can work properly.

6.2 System Information for the NVP-300 series.

- When you login the web page, you can see the NVP-300 series current system information like firmware version, company… etc in this page.
- Also you can see the function lists in the left side. You can use mouse to click the function you want to set up.


System Information
This page illustrate the system related information.
Model Name: VoIP
Firmware Version: Wed Oct 12 17:08:27 2005.
Codec Version: Fri Oct 14 17:07:38 2005.

6.3 Phone Book

The NVP-300 series supports user programming of up to ten phone number, local,international or emergency numbers and/or IP addresses for fast and easy access.

6.3.1 In Phone Book contains Speed Dial Settings. You can setup the Speed Dial number. If you want to use Speed Dial you just dial the speed dial number (from 0~9) then press "#".

6.3.2 In Speed Dial setting function you can add/delete Speed Dial number. You can input maximum 10 entries speed dial list.

- If you need to add a phone number into the Speed Dial list, you need to input the position, the name, and the phone number (by URL type). When you finished a new phone list, just click the "Add Phone" button.

- If you want to delete a phone number, you can select the phone number you want to delete then click "Delete Selected" button.
- If you want to delete all phone numbers, you can click "Delete All" button.

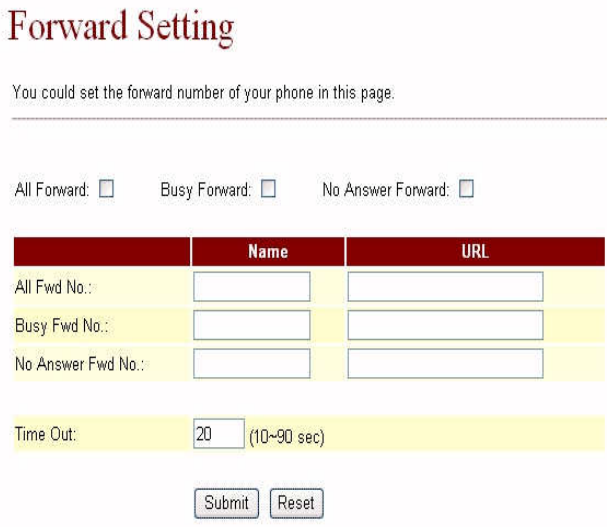
6.4 Phone Setting

In Phone Setting contains Call Forward, SNTP Settings, Volume Settings, Block Setting, Caller ID, Auto Dial Setting, Flash Time Setting and Call Waiting Setting functions. Please configure the items as follows.

6.4.1 Call Forward function:

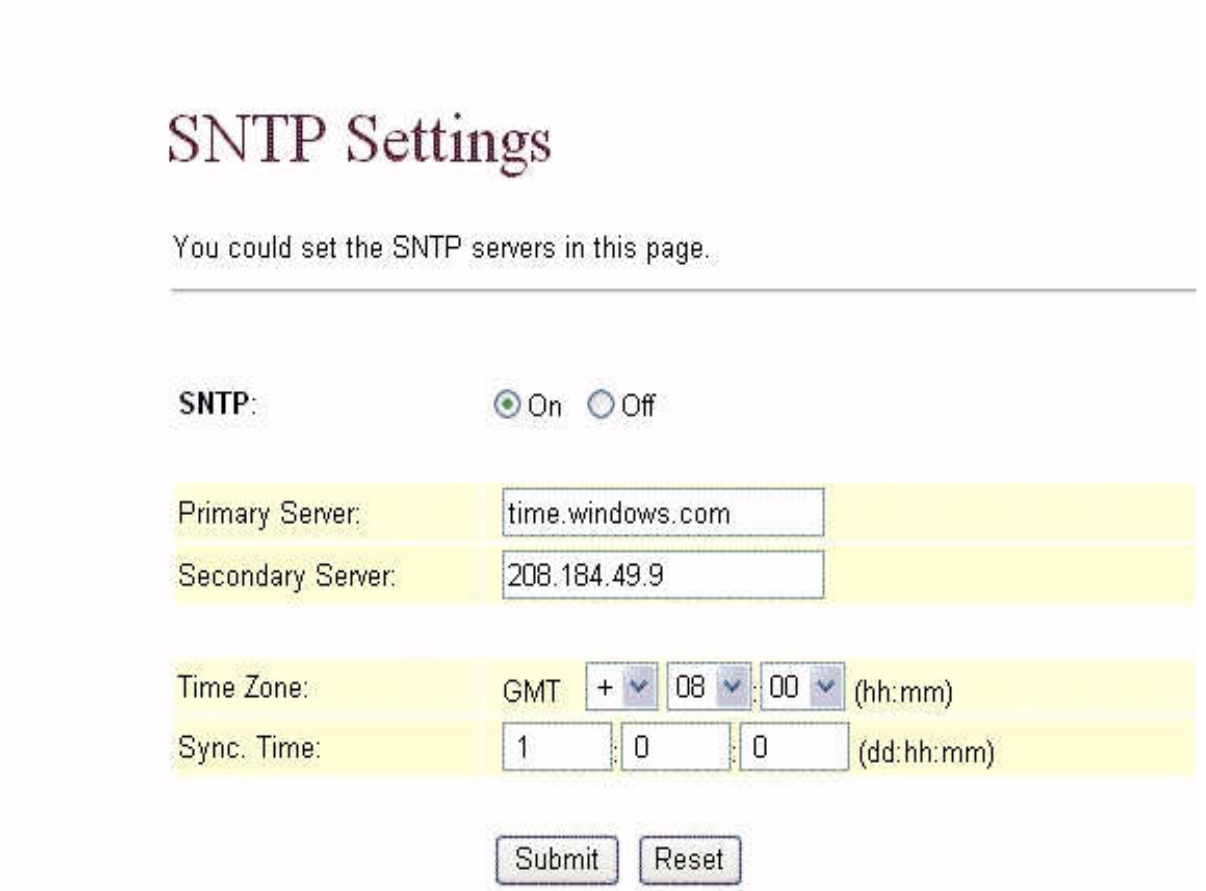
you can setup the phone number you want to forward in this page.There are three type of Forward mode. You can choose All Forward, Busy Forward, and No Answer Forward by click the icon.

- All Forward: All incoming call will forward to the number you choosed. You can input the name and the phone number in URL field. If you select this function, then all the incoming call will direct forward to the speed dial number you choose.
- Busy Forward: If you are on the phone, the new incoming call will forward to the number you choosed. You can input the name and the phone number in URL field.
- No Answer Forward: : If you can not answer the phone, the incoming call will forward to the number you choosed. You can input the name and the phone number in URL field. Also you have to set the Time Out time for system to start to forward the call to the number you choosed.
- 6.4.2.4 When you finished the setting, please click the Submit button.



6.4.2 SNTP Setting function:

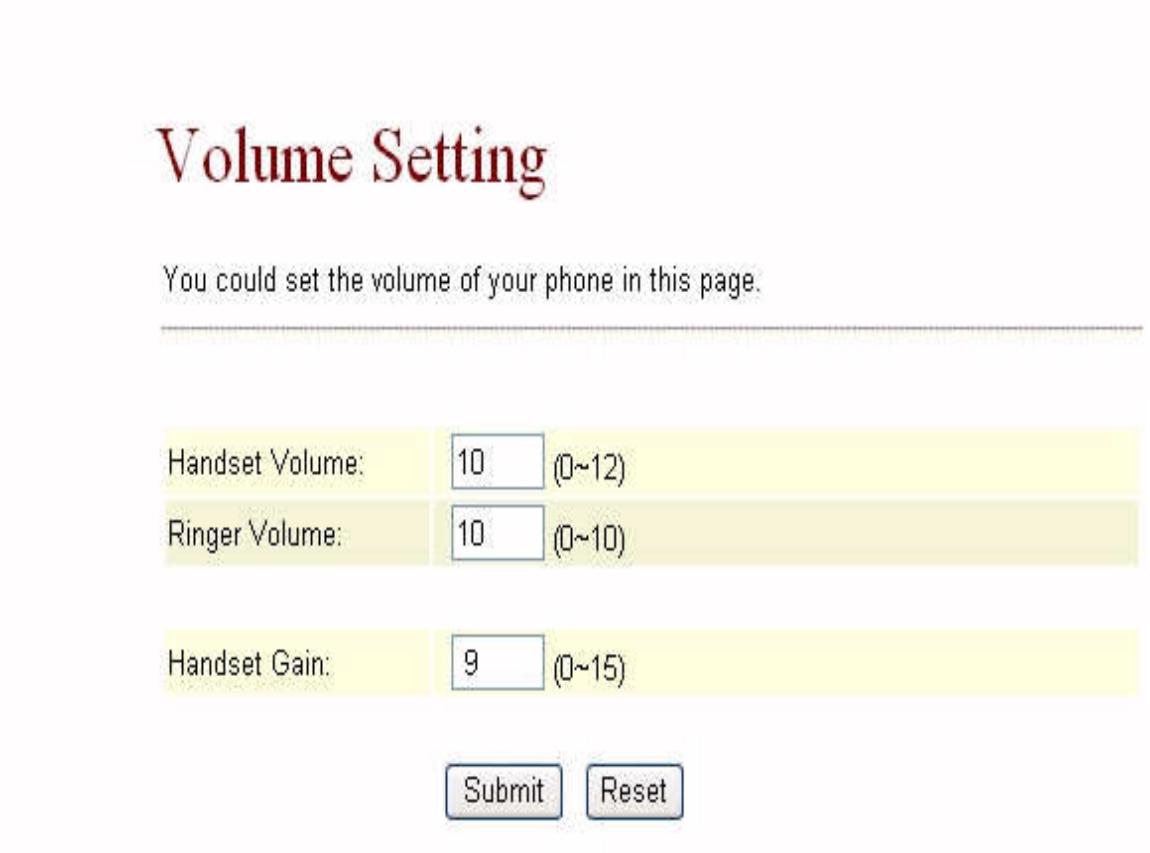
you can setup the primary and second SNTP Server IP Address, to get the date/time information. Also you can base on your location to set the Time Zone, and how long need to synchronize again. When you finished the setting, please click the Submit button.



6.4.3 Volume Setting function:

you can setup the Handset Volume, Ringer Volume, and the Handset Gain. When you finished the setting, please click the Submit button.

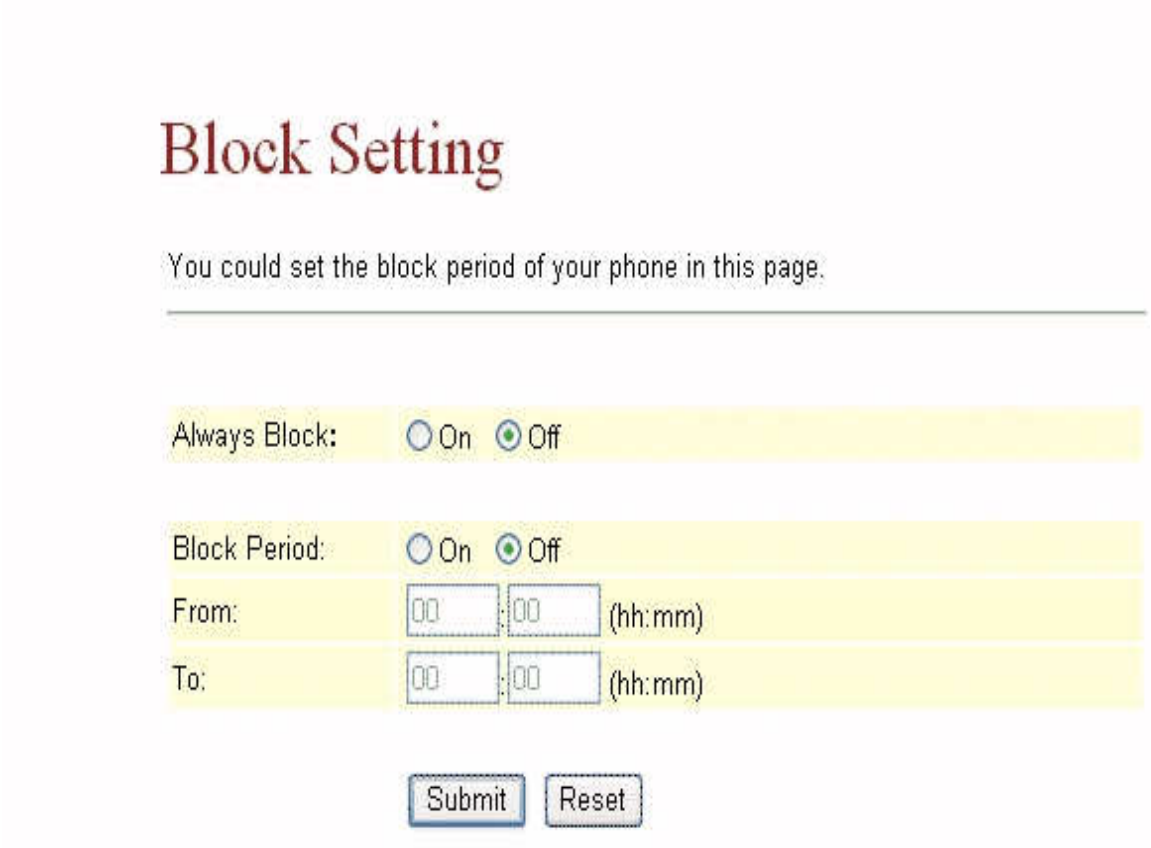
- Handset Volume is to set the volume for you can hear from the handset.
- Ringer Volume is to set the ringer volume for you can hear.
- PSTN-Out Volume is to set the PSTN volume for you can hear.
- Handset Gain is to set the volume send out to the other side's handset.
- PSTN-In Gain is to set the volume send out to the other side's handset.



6.4.4 Block Setting function:

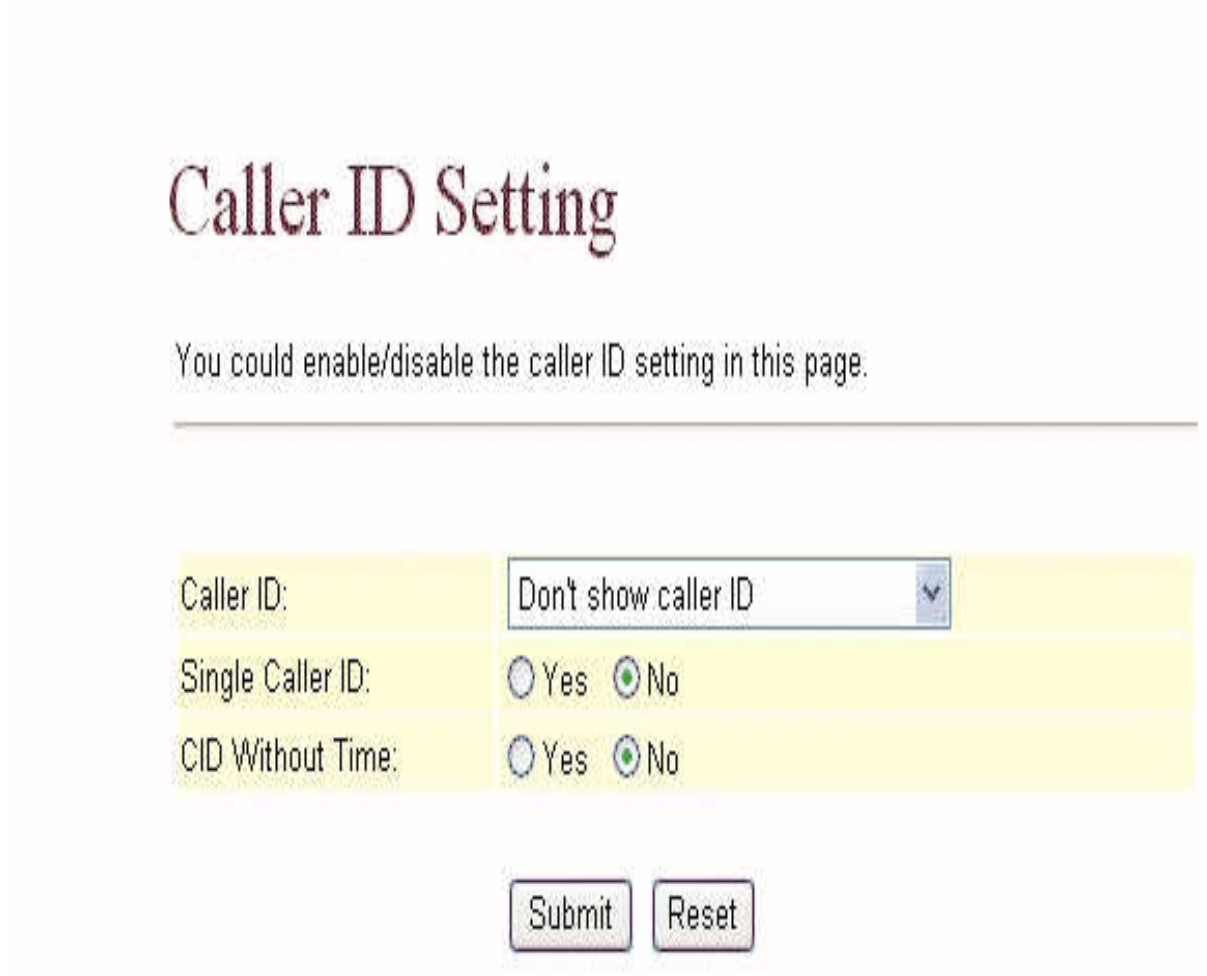
you can setup the Block Setting to keep the phone slience. You can choose Always Block or Block a period.

- Always Block: All incoming call will be blocked until disable this feature.
- Block Period: Set a time period and the phone will be blocked during the time period. If the "From" time is large than the "To" time, the Block time will from Day 1 to Day 2.
- When you finished the setting, please click the Submit button.



6.4.5 Caller ID function:

you can set the device to show Caller ID in your PSTN Phone or IP Phone. There are four selection of Caller ID. You need to base on your environment to set the Caller ID function for FSK or DTMF.

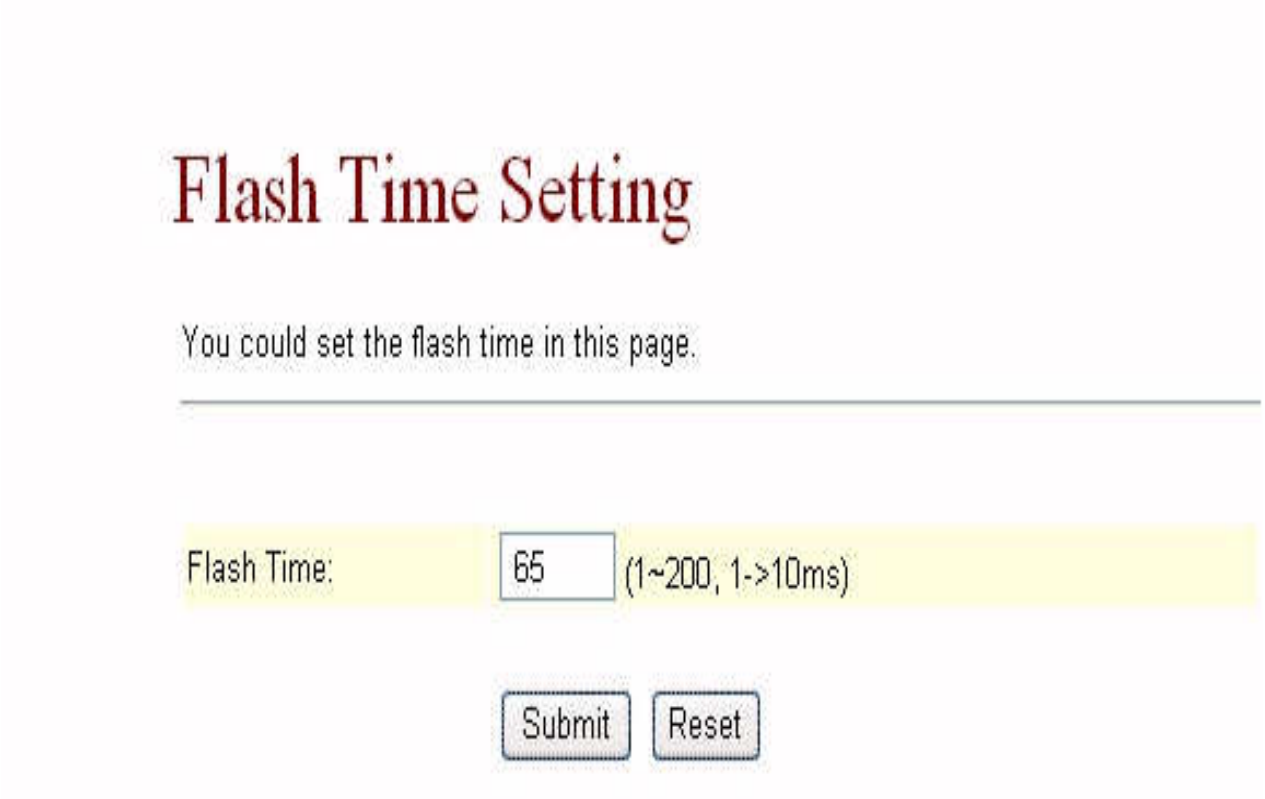


6.4.6 Auto Dial Setting function:

This function is when you input the phone number by the keypad but you don't need to press "#". After time out the system will dial directly.

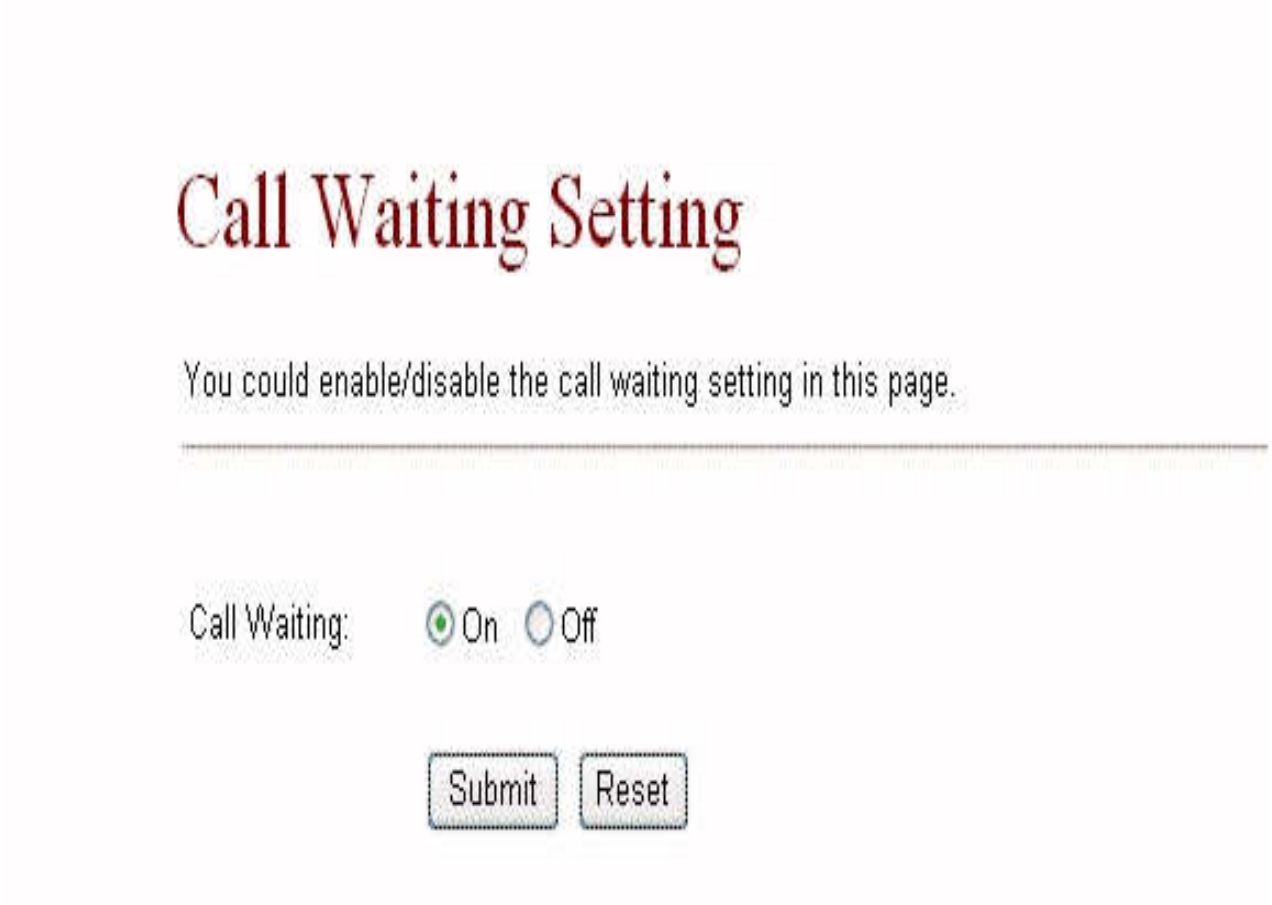
6.4.7 Flash Time Setting function:

When you use the PSTN Phone and you need to press the Hook to do the Flash (Switch to the other phone line or HOLD), this function is for you to set the time you press the Hook to represent the Flash function.



6.4.8 Call Waiting Setting function:

You can Enable/Disable the Call Waiting function, When you are talking with someone, there is a new incoming call, you will hear the call waiting tone.



6.4.9 T.38 Setting function:

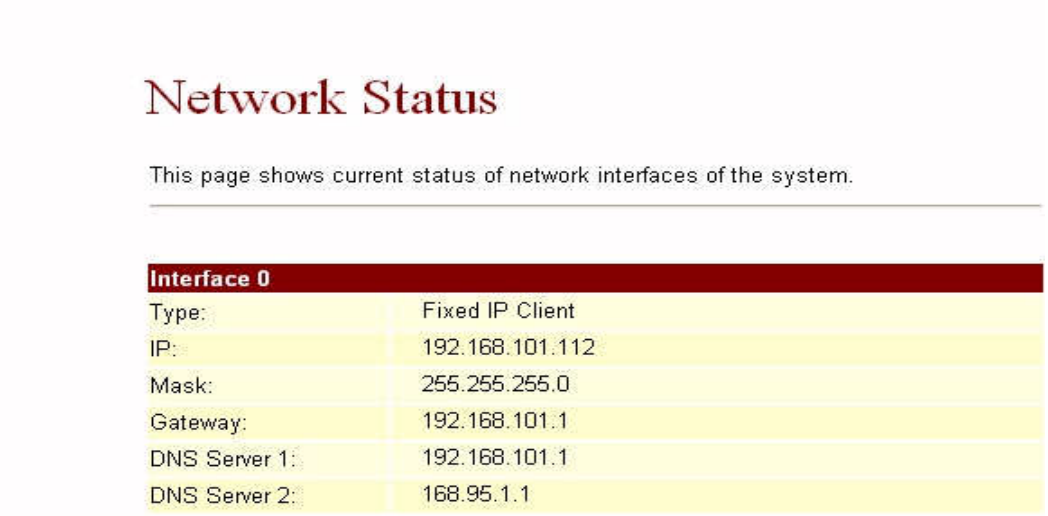
You can Enable/Disable the T.38 function.

6.5 Network

In Network you can check the Network status, configure the Network Settings and DDNS settings.

6.5.1 Network Status:

You can check the current Network setting in this page.



6.5.2 WAN Settings:

In this page you can configure the IP Phone WAN port's setting. The WAN port is for you to connect to the ADSL Router, Broadband Router. Also you can use PPPoE to get the WAN IP address from your ISP.

- The IP Phone's default setting is NAT mode. If you don't need to use the NAT Mode, you can chang to Bridge Mode. If you change the setting to Bridge Mode, then the LAN setting will not effect and will be the same as WAN port.
- The WAN port default is DHCP Client mode, You can change the setting to Fixed IP Mode, or PPPoE Mode.
- If you change the WAN port's setting to Fix IP Mode, then you have to make sure the IP address. Net Mask, Gateway, and DNS setting is suitable in your current network environment.
- If you change the WAN port's setting to PPPoE Mode, you have to input a correct username/password to get the IP address from your Internet Service Provider.
- When you finished the setting, please click the Submit button.
- If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.

WAN Settings

You could configure the WAN settings in this page.

LAN Mode:	○ Bridge ◉ NAT

WAN Setting

IP Type:	○ Fixed IP ◉ DHCP Client ○ PPPoE
IP:	61.216.231.217
Mask:	255.0.0.0
Gateway:	61.216.116.254
DNS Server1:	168.95.192.1
DNS Server2:	168.95.1.1
MAC:	fc45315284a2

PPPoE Setting

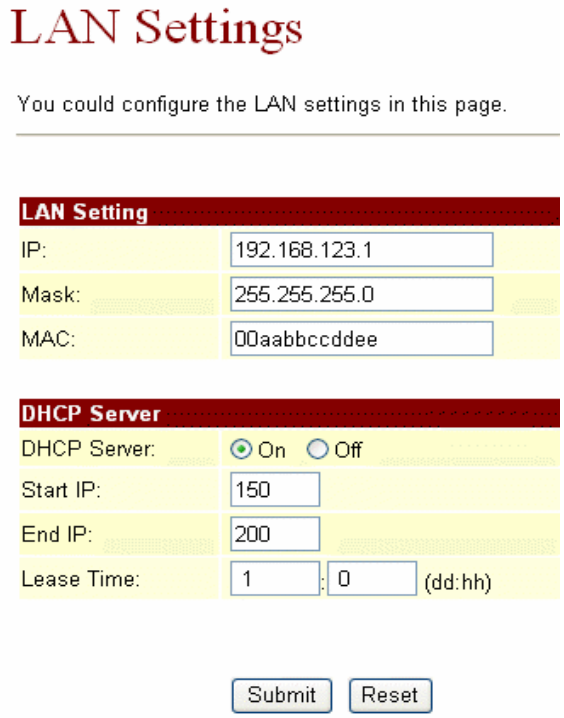
User Name:	
Password:	

Submit Reset

6.5.3 LAN Settings:

In this page you can configure the IP Phone LAN port's setting.

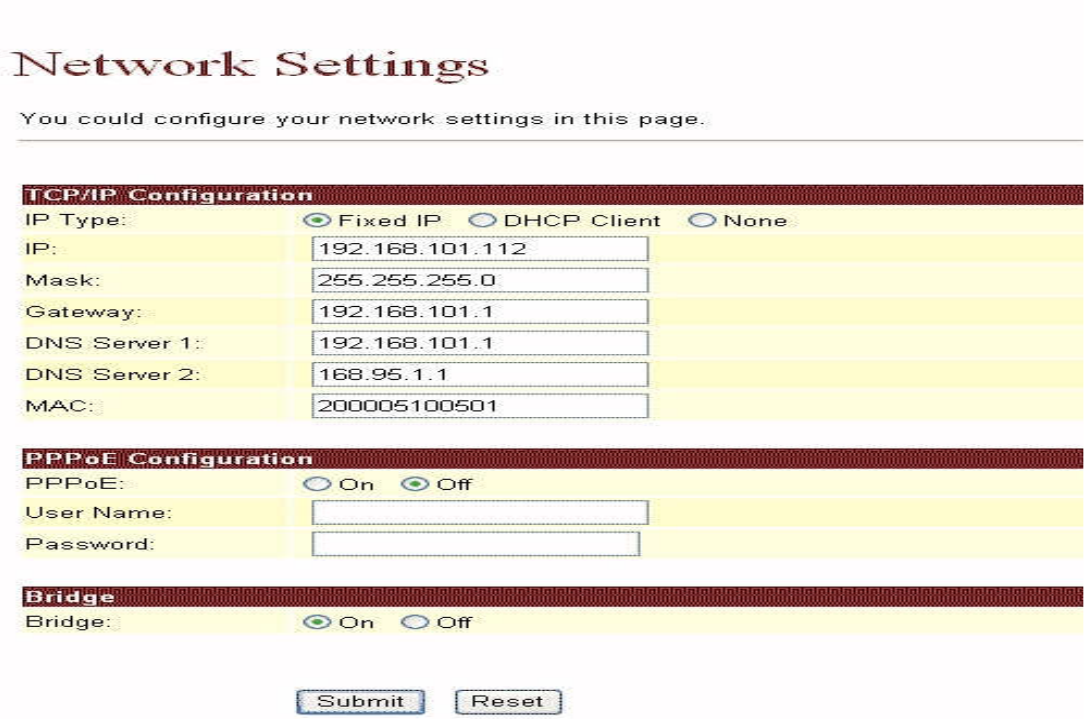
- The LAN port's default IP address is 192.168.123.1, Net Mask is 255.255.255.0., and DHCP Server enabled. The start IP address if 150, end IP adress is 200. It is not necessary to change the LAN settings.
- You can connect your PC to the LAN port, set your PC as DHCP Client mode, then you can get IP addreess from the NVP-300.
- When you finished the setting, please click the Submit button.
- If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.



6.5.4 Network Settings:

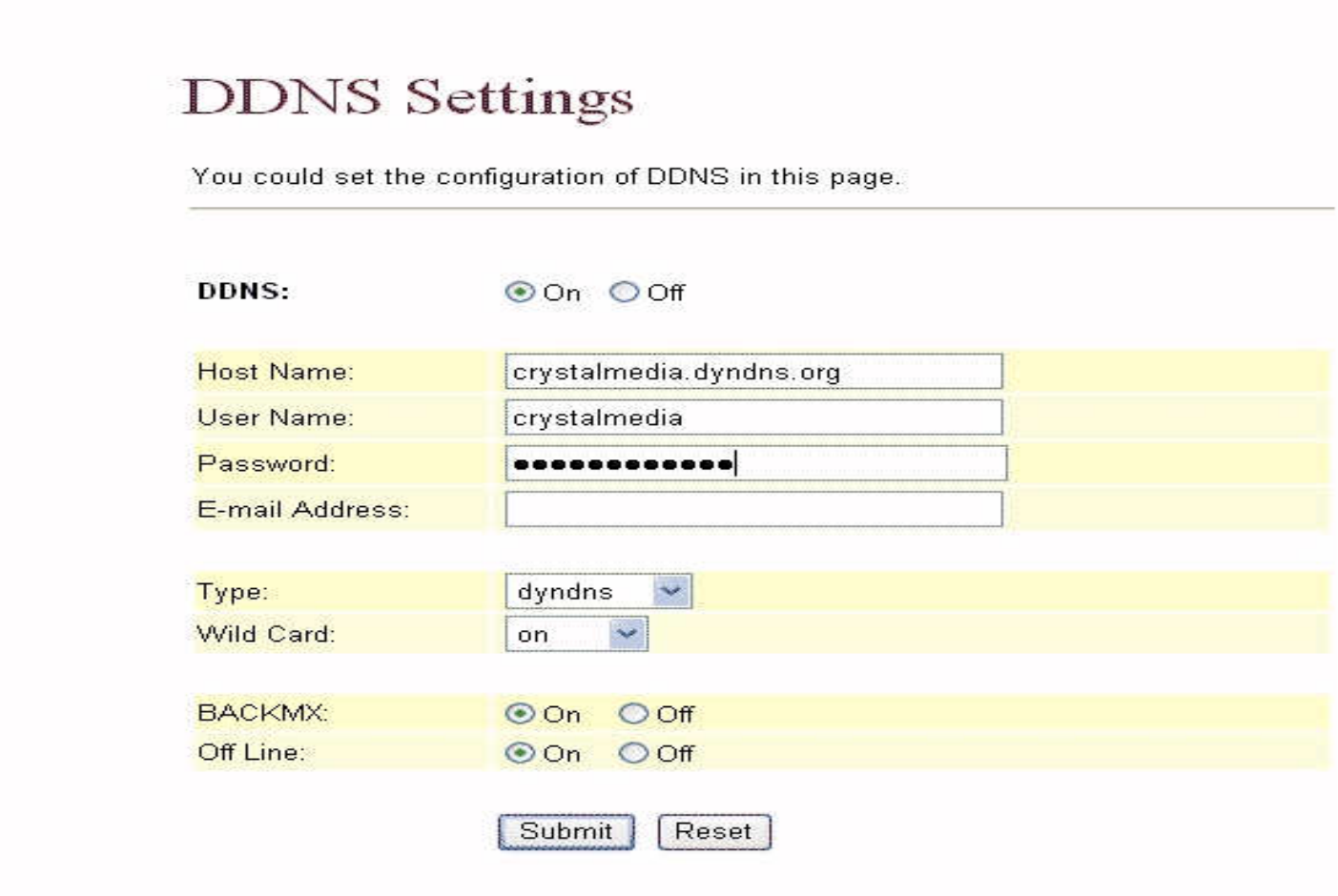
You can configure the NVP-300 series Network setting in this page.

- The TCP/IP Configuration item is to setup the LAN port's network environment. You may refer to your current network environment to configure the NVP-300 series properly.
- The PPPoE Configuration item is to setup the PPPoE Username and Password. If you have the PPPoE account from your Service Provider, please input the Username and the Password correctly.
- The Bridge Item is to setup the NVP-300 series Bridge mode Enable/Disable. If you set the Bridge On, then the two Fast Ethernet ports will be transparent.
- When you finished the setting, please click the Submit button.



6.5.5 DDNS Setting:

You can configure the DDNS setting in this page. You need to have the DDNS account and input the informations properly. You can have a DDNS account with a public IP address then others can call you via the DDNS account. But now most of the VoIP applications are work with a SIP Proxy Server. When you finished the setting, please click the Submit button.



6.5.6 VLAN Setting:

You can set the VLAN setting in this page. There are two parts in this page. First one is to set the packets related to the NVP-300, and the second parts is if you use the VLAN setting in the NAT Mode.

- There are two kind of destination packets will come from the VoIP's WAN port, one kind of packets will go to the VoIP, the other will go through the LAN port to the PC.
- VLAN Packets: if you enable the first VLAN Packets and set the VID, User Priority, and CFI, then all the incoming packets will be check with the IP Address and the VID.
- **VID:** You can follow your service provider to set your VID.
- User Priority: Defines user priority, giving eight (2^3) priority levels. IEEE 802.1P defines the operation for these 3 user priority bits. Usually this will be defined by your service provider.
- **CFI:** Canonical Format Indicator is always set to zero for Ethernet switches. CFI is used for compatibility reason between Ethernet type network and Token Ring type network. If a frame received at an Ethernet port has a CFI set to 1, then that frame should not be forwarded as it is to an untagged port.
- When you enable the first VLAN Packets and set the VID, User Priority, and CFI, then all the incoming packets with the VoIP's IP address and the same VID will be accept by the VoIP. If the incoming packets with the VoIP's IP address but the different VID then the packets will be discard by the VoIP. The Other incoming packets with different IP address will go through the LAN port to the PC.
- **NAT VLAN Setting:** When you set your device in NAT mode, the VoIP can help you to filter the wrong incoming packets. You can separate the other device connectd behind the VoIP into 4 VLAN group. You can set different VID for these 4 groups. When the incoming packets go through the VoIP's WAN port then the VoIP will check the VID, if the packets is not going to the VoIP(with the VoIP's IP address and the correct VID), and the VID is not these four VID you set, then the packets will be discard by the VoIP.
- If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.

Note:

NVP-300 series only provide one LAN port, so V-LAN function can't be setted.

VLAN Settings

You could set the VLAN settings in this page.

VLAN Packets:	(•) Off ( ) On
VID:	136 (2 ~ 4094)
User Priority:	0 (0 ~ 7)
CFI:	0 (0 ~ 1)

NAT VLAN Setting

VLAN Packets:	(•) Off ( ) On
VID1:	4 (2 ~ 4094), 0->Off
VID2:	5 (2 ~ 4094), 0->Off
VID3:	6 (2 ~ 4094), 0->Off
VID4:	7 (2 ~ 4094), 0->Off

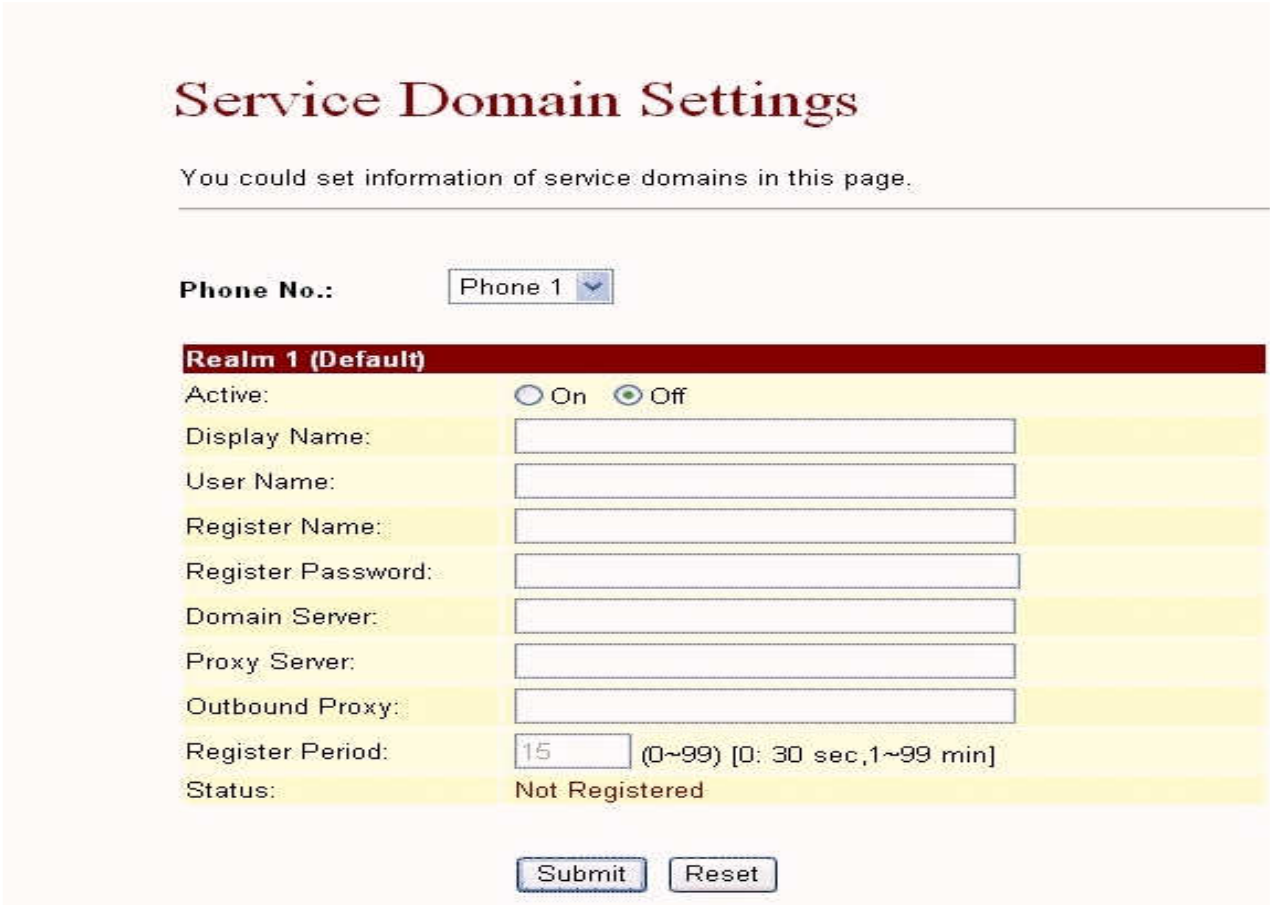
Submit Reset

6.6 SIP Settings

In SIP Settings you can setup the Service Domain, Port Settngs, Codec Settings, Codec ID Setting, RTP Setting, RPort Setting and Other Settings. If the VoIP service is provided by ISP, you need to setup the related informations correctly then you can register to the SIP Proxy Server correctly.

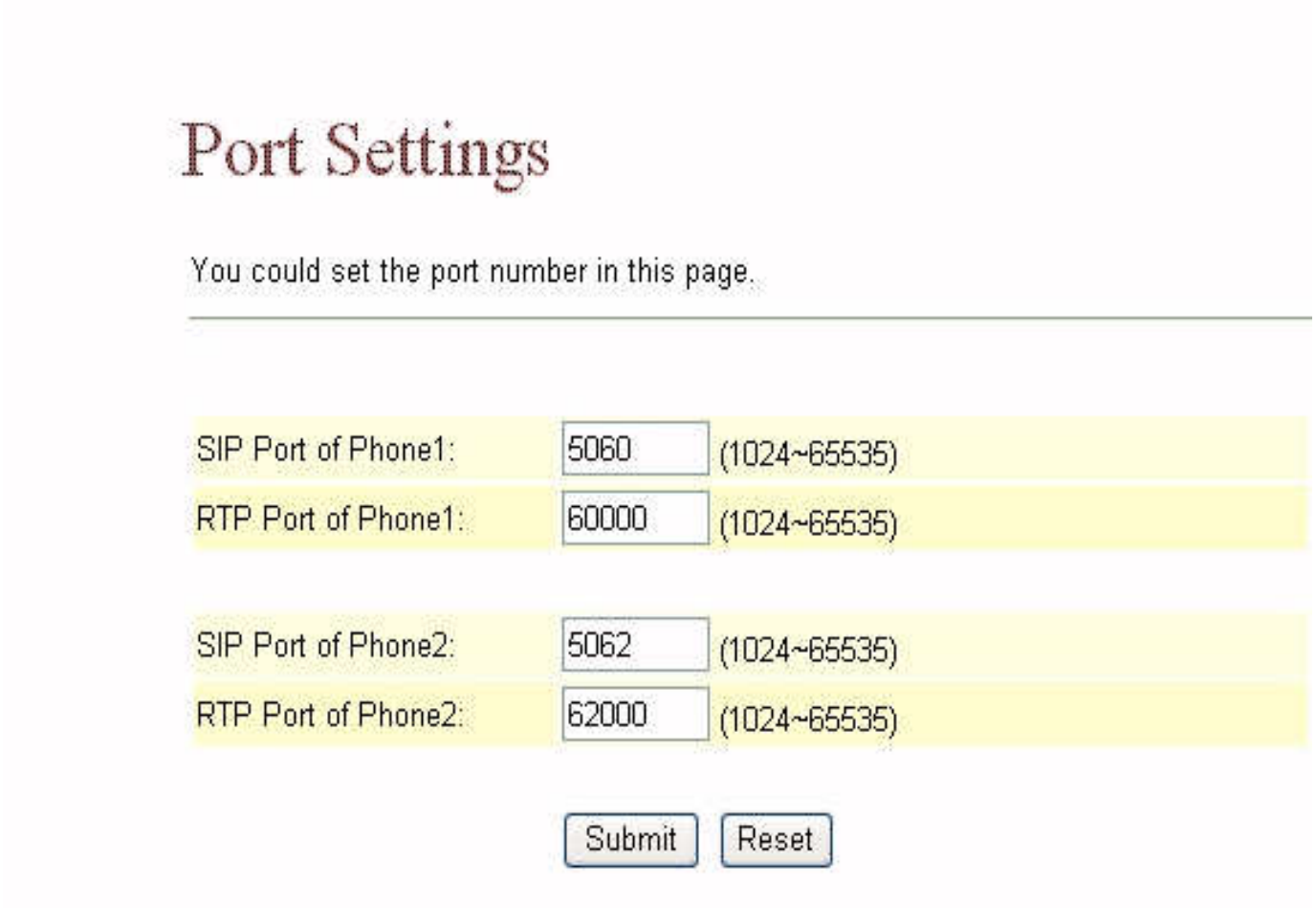
In Service Domain Function you need to input the account and the related informations in this page, please refer to your ISP provider. You can register three SIP account in the NVP-300 series. You can dial the VoIP phone to your friends via first enable SIP account and receive the phone from these three SIP accounts. For the second phone you can use the same way to register.

- **Display Name:** you can input the name you want to display.
- **User Name:** you need to input the User Name get from your ISP.
- **Register Name:** you need to input the Register Name get from your ISP.
- **Register Password:** you need to input the Register Password get from your ISP.
- **Domain Server:** you need to input the Domain Server get from your ISP.
- **Proxy Server:** you need to input the Proxy Server get from your ISP.
- **Outbound Proxy:** you need to input the Outbound Proxy get from your ISP. If your ISP does not provide the information, then you can skip this item.
- **Register Period:** you need to input the Register Period get from your ISP. This is count in minute.
- You can see the Register Status in the Status item. If the item shows "Registered", then your NVP-300 series is registered to the ISP, you can make a phone call direcly.If you have more than one SIP account, you can following the steps to register to the other ISP.
- When you finished the setting, please click the Submit button.


Service Domain Settings
You could set information of service domains in this page.
Phone No.:
Phone 1
Realm 1 (Default)
Active:
On
Off
Display Name:
User Name:
Register Name:
Register Password:
Domain Server:
Proxy Server:
Outbound Proxy:
Register Period:
15
(0~99) [0: 30 sec,1~99 min]
Status:
Not Registered
Submit
Reset

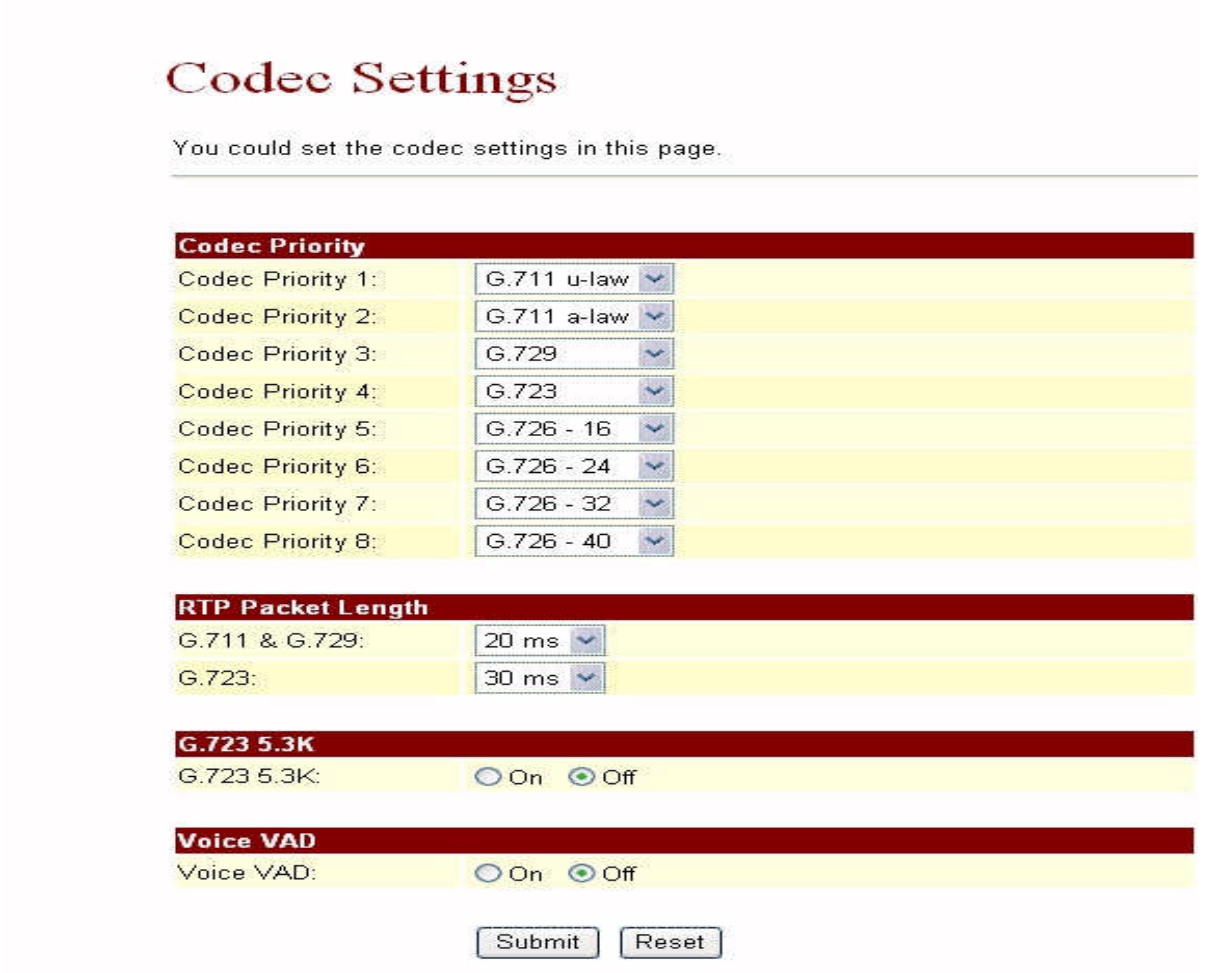
6.6.1 Port Settings:

you can setup the SIP and RTP port number in this page. Each ISP provider will have different SIP/RTPport setting, please refer to the ISP to setup the port number correctly. When you finished the setting, please click the Submit button.



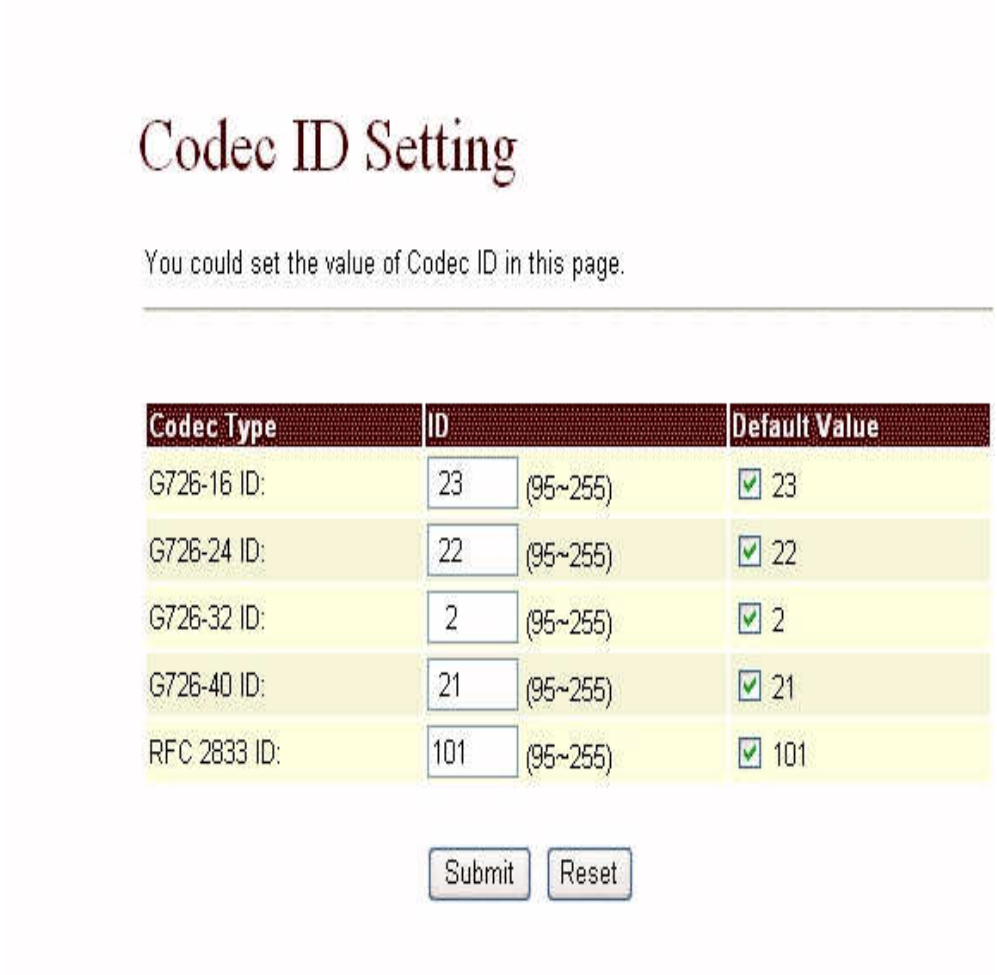
6.6.2 Codec Settings:

you can setup the Codec priority, RTP packet length, and VAD function in this page. You need to follow the ISP suggestion to setup these items. When you finished the setting, please click the Submit button.


Codec Settings
You could set the codec settings in this page.
Codec Priority
Codec Priority 1: G.711 u-law
Codec Priority 2: G.711 a-law
Codec Priority 3: G.729
Codec Priority 4: G.723
Codec Priority 5: G.726 - 16
Codec Priority 6: G.726 - 24
Codec Priority 7: G.726 - 32
Codec Priority 8: G.726 - 40
RTP Packet Length
G.711 & G.729: 20 ms
G.723: 30 ms
G.723 5.3K
G.723 5.3K: On Off
Voice VAD
Voice VAD: On Off
Submit
Reset

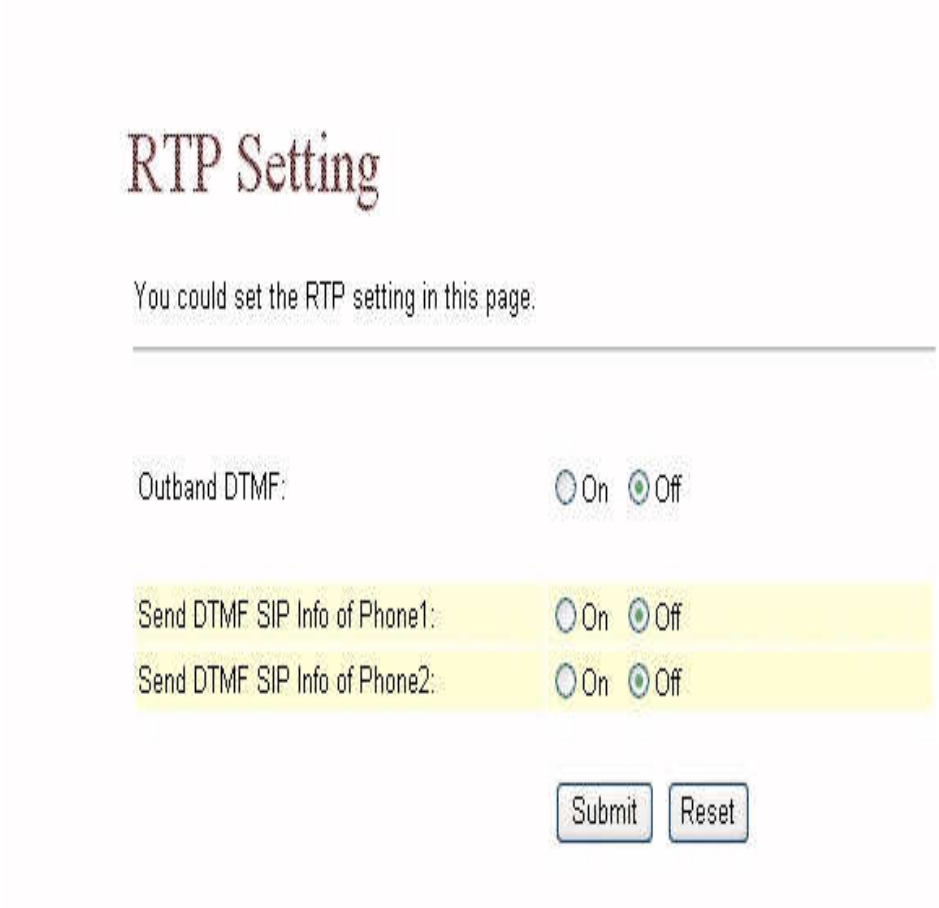
6.6.3 Codec ID Setting:

Sometimes 2 VoIP device with different Codec ID will cause the interopability issue. If you are talking with others got some problems, you may ask the other one what kind of Codec ID he use, then you can change your Codec ID. When you finished the setting, please click the Submit button.



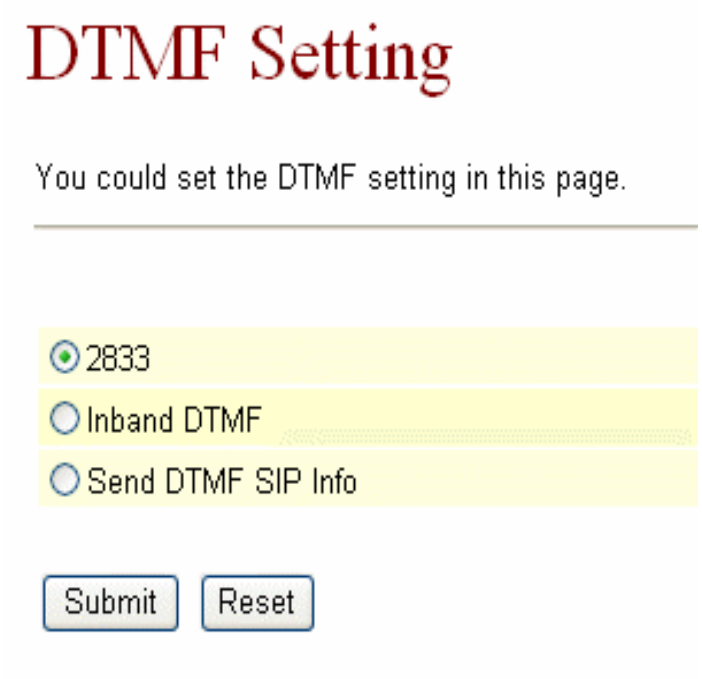
6.6.4 RTP Setting:

you can setup the Out-Band DTMF and send DTMF SIP Info Enable/Disable in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.



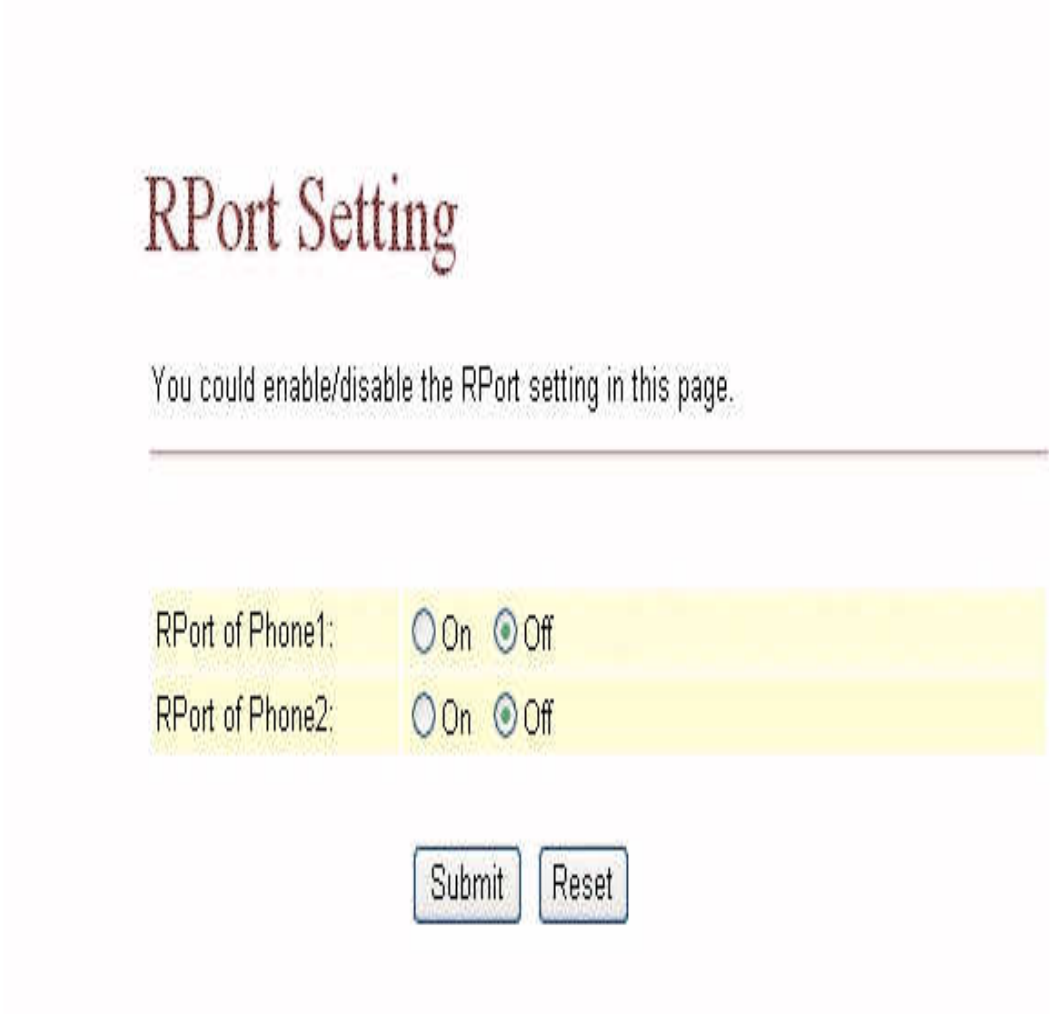
6.6.5 DTMF Setting:

you can setup the InBand DTMF, 2833 Out-Band DTMF and Send DTMF SIP Info Enable/Disable in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.



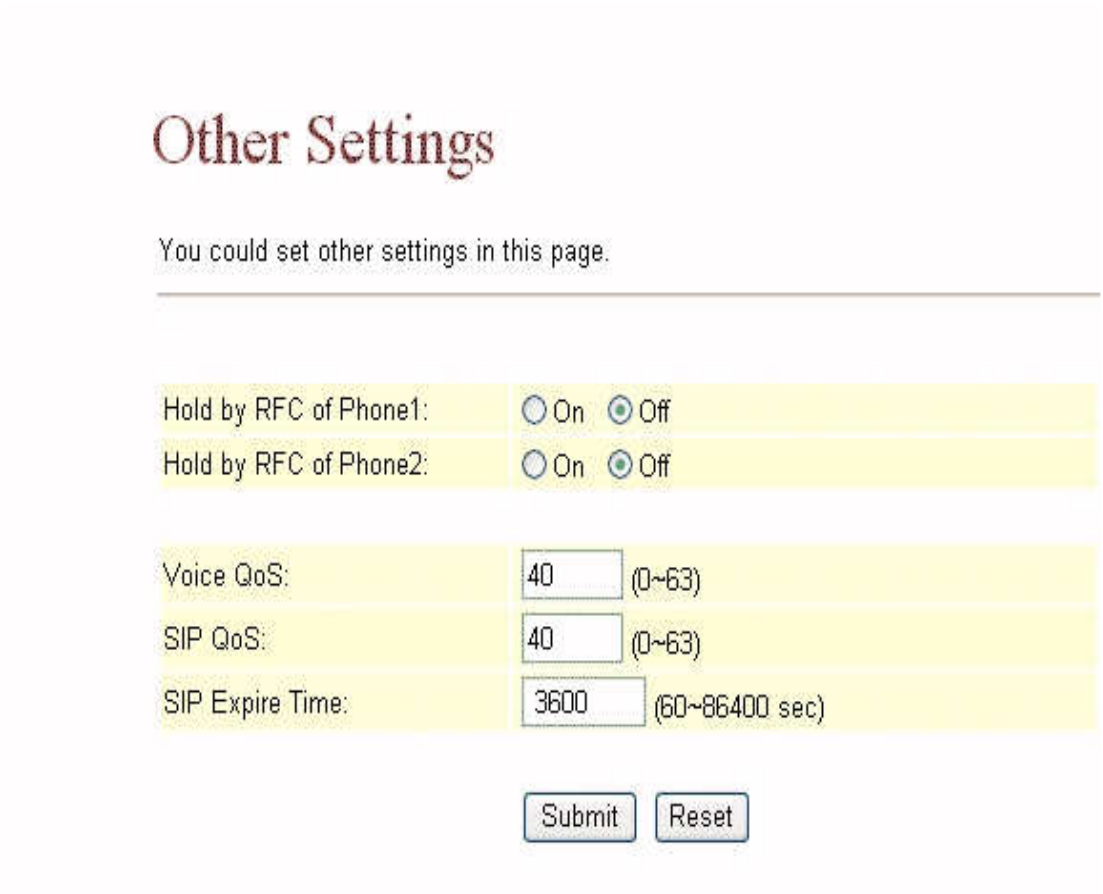
6.6.6 RPort Function:

you can setup the RPort Enable/Disable in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.



6.6.7 Other Settings:

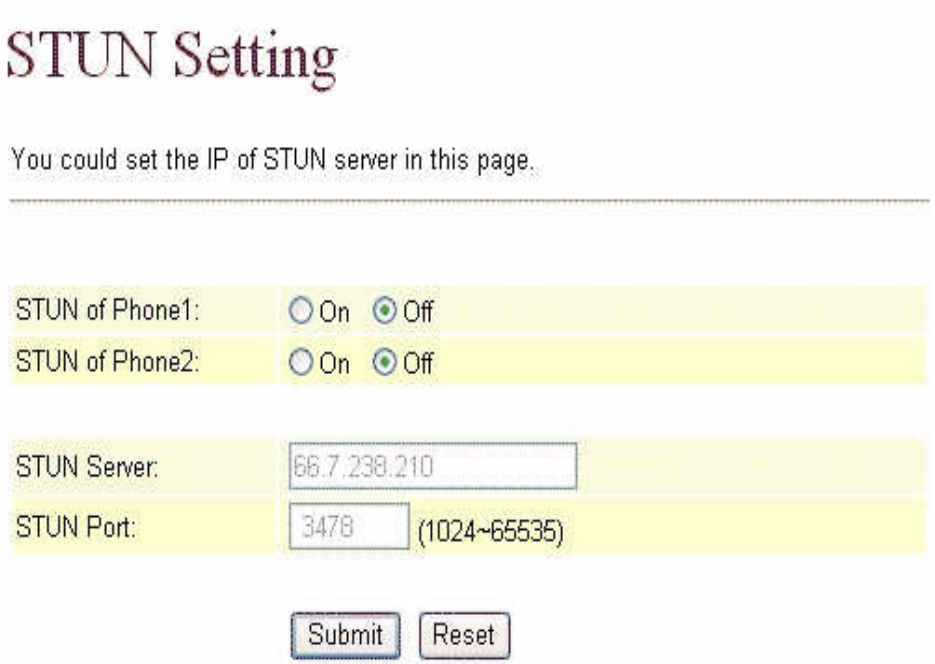
you can setup the Hold by RFC, Voice/SIP QoS and SIP expire time in this page. To change these settings please following your ISP information. When you finished the setting, please click the Submit button. The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice packets will get the higher priority to the Internet. But the QoS function still need to cooperate with the others Internet devices.



6.7 NAT Trans.

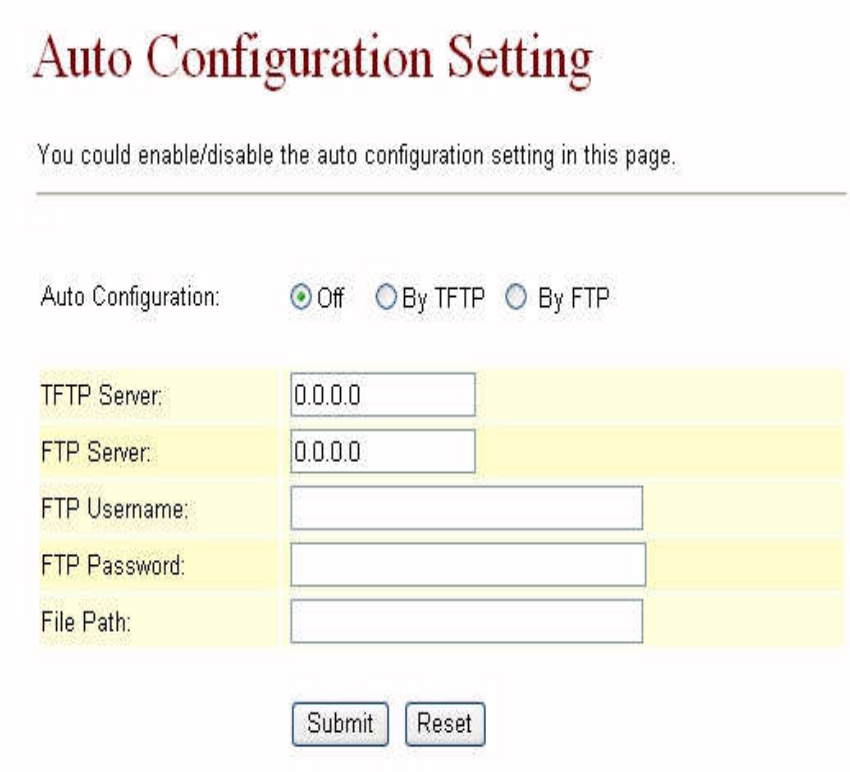
In NAT Trans:
You can setup STUN function. These functions can help your NVP-300 series working properly behind NAT.

STUN Setting:
You can setup the STUN Enable/Disable and STUN Server IP address in this page. This function can help your NVP-300 series working properly behind NAT. To change these settings please following your ISP information. When you finished the setting, please click the Submit button.

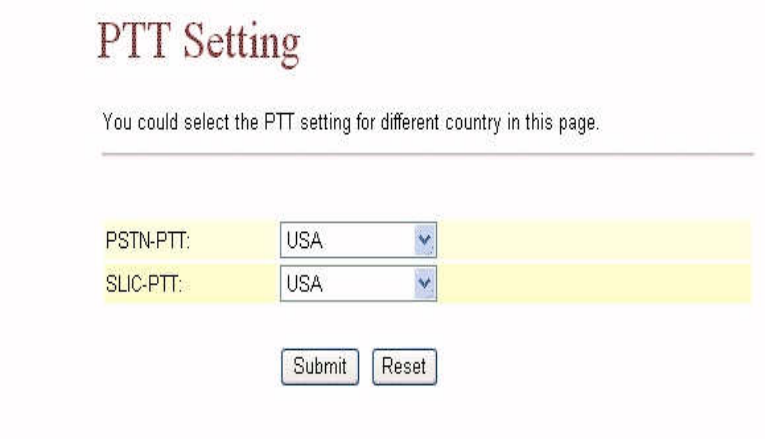


6.8 Others

- In Others you can setup Auto Config, PTT Setting and ICMP Setting function. The function can configure your VoIP Phone automatically.
- **Auto Config:** you can setup the Auto Configuration Enable/Disable and auto configuration by FTP or TFTP. You need to select the way to do the Auto Configuration and set the Server IP address in this page. This function can automatically download the configure file to setup your NVP-300. When you finished the setting, please click the Submit button.



- **PTT Setting:** You can setup the PTT in this page. When you are using different country's PSTN Phone or connect to different country's PSTN Line, you have to set the country's setting to meet the requirement. When you finished the setting, please click the Submit button.



- **ICMP Setting:** You can setup the ICMP echo Enable/Disable in this page. This function can disable echo when someone ping this device, it can avoid haker try to attack the device. When you finished the setting, please click the Submit button.

6.9 System Auth.

In System Authority you can change your login name and password.

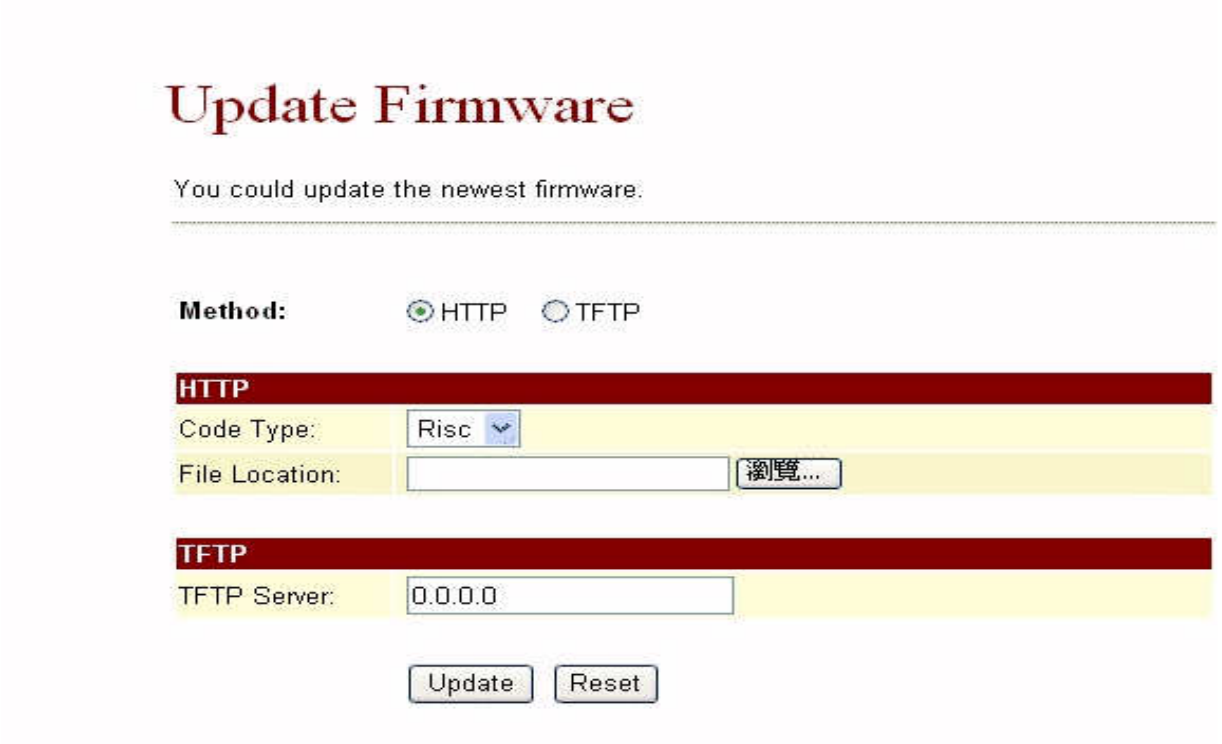
6.10 Save Change

In Save Change you can save the changes you have done. If you want to use new setting in the NVP-300 series, You have to click the Save button. After you click the Save button, the NVP-300 series will automatically restart and the new setting will effect.

6.11 Update

- In Update you can update the NVP-300 series's firmware to the new one or do the factory reset to let the NVP-300 series back to default setting.
- In New Firmware function you can update new firmware via HTTP in this page. You can ugrade the firmware by the following steps:

1. Select the firmware code type, Risc or DSP code.
2. Click the "Browse" button in the right side of the File Location or you can type the correct path and the filename in File Location blank.
3. Select the correct file you want to download to the NVP-300 series then click the Update button.



- In Default Setting you can restore the NVP-300 series to factory default in this page. You can just click the Restore button, then the NVP-300 series will restore to default and automatically restart again.

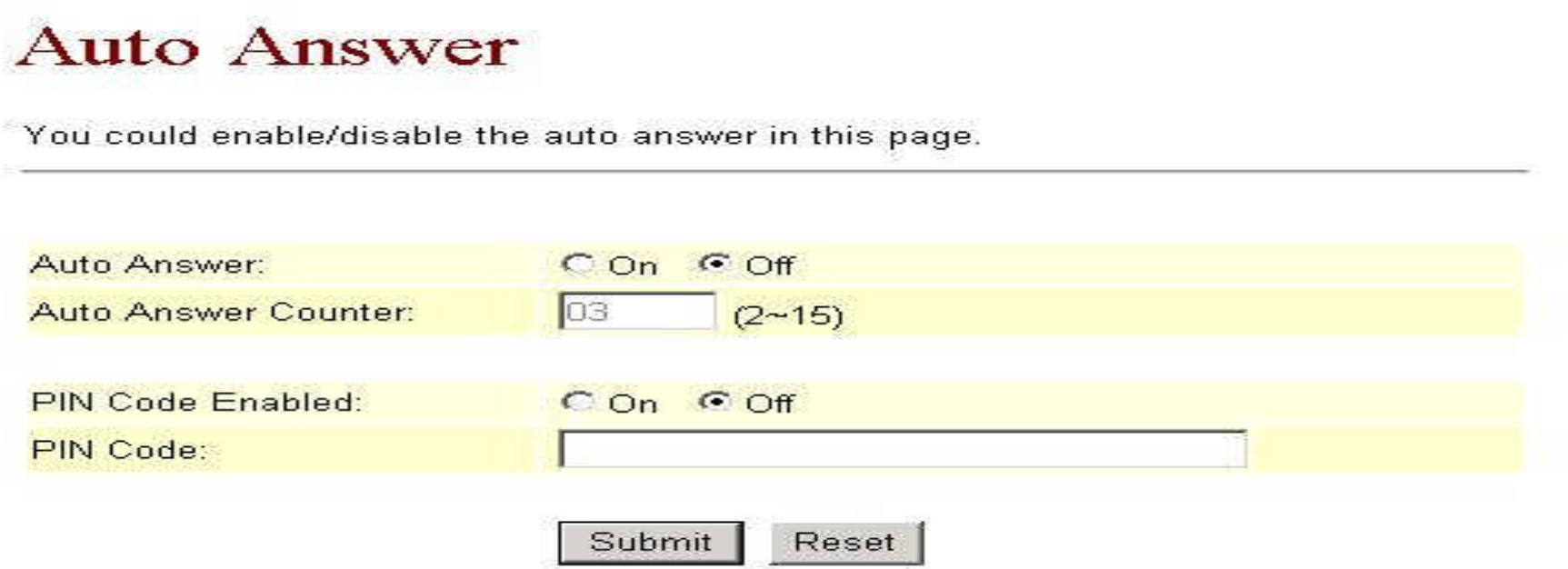
Restore Default Settings
You could click the restore button to restore the factory settings.
Restore default settings:
Restore

6.12 Reboot

Reboot function you can restart the NVP-300 series. If you want to restart the NVP-300 series, you can just click the Reboot button, then the NVP-300 series will automatically.

6.13 Auto Answer(NVP-300SO/SSO only)

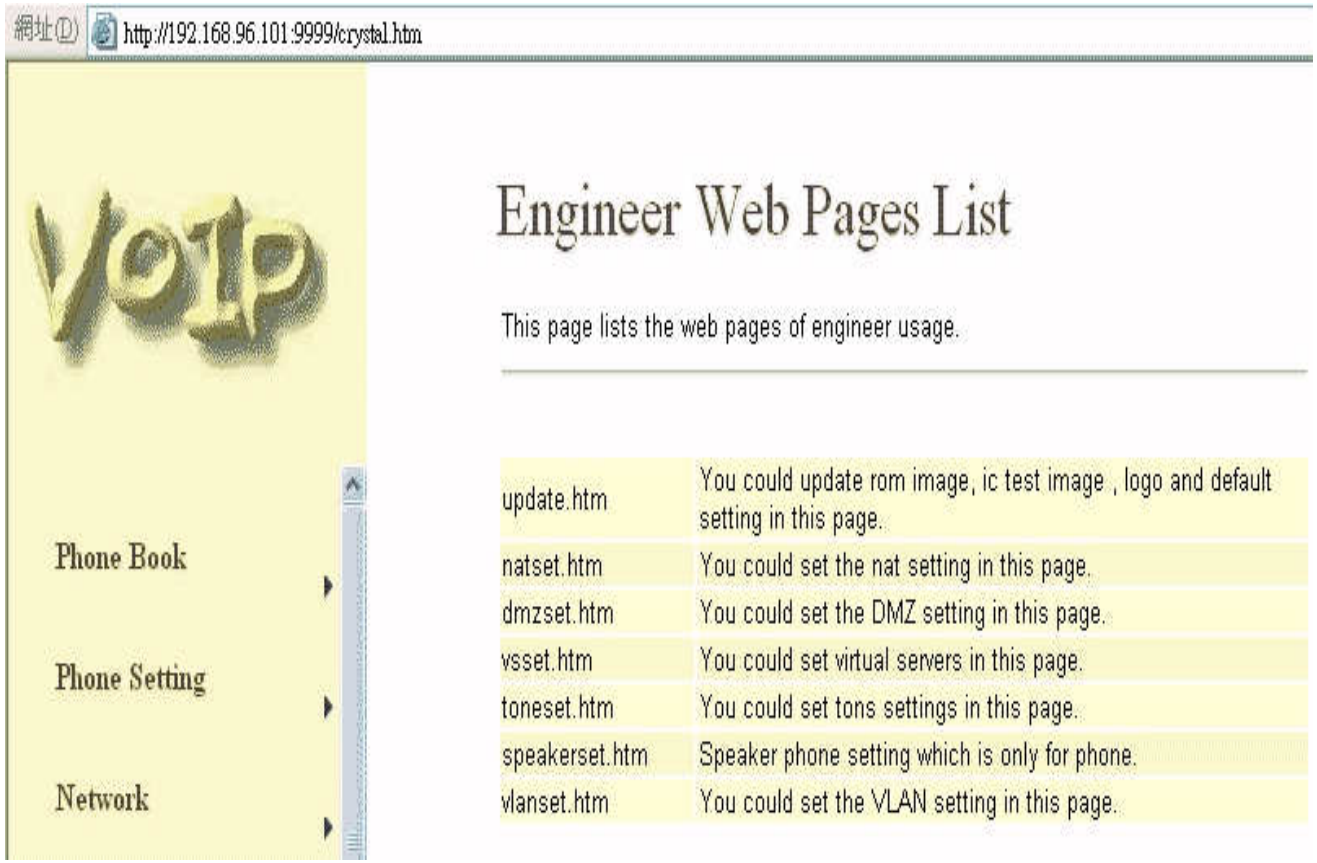
- Auto Answer will auto change from FXS to FXO(PBX) when call in without reply over setting time. Call in persons can call you again through FXO when NVP-300 series auto change to FXO and hearing a dial tone.
- Plese select ON, If you want to enable Auto Answer function on the NVP-300 series, and setting pin code(password) when you only provide this function to you well know persons.



Chapter 7. Engineering webpage

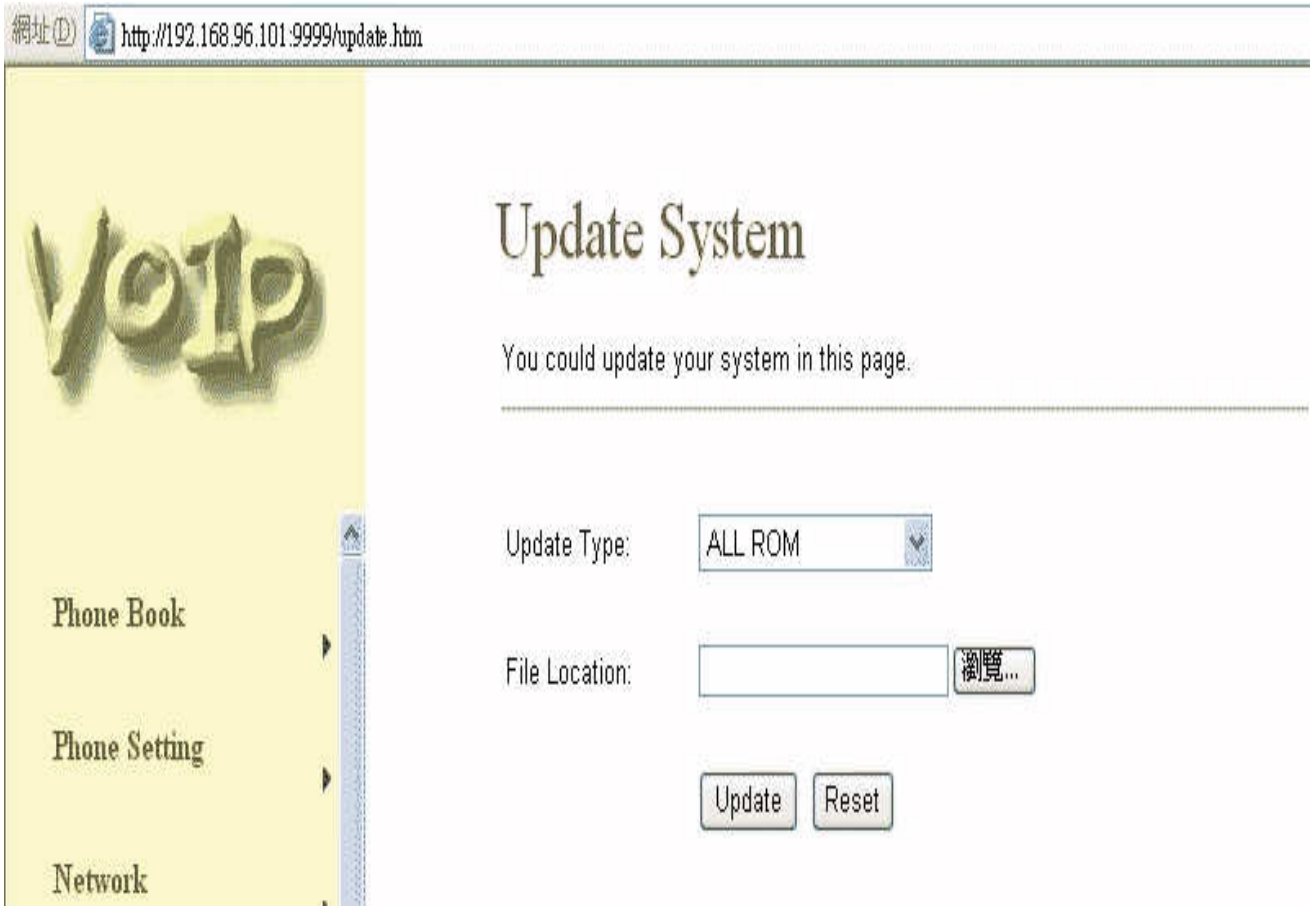
7.1 Engineer usage webpage list

You have to login the system first then change the webpage to crystal.htm manually. In this webpage you will see the list about engineer webpage. You can change the webpage to what you want.


網址(D)
http://192.168.96.101:9999/crystal.htm
VoIP
Phone Book
Phone Setting
Network
Engineer Web Pages List
This page lists the web pages of engineer usage.
update.htm
You could update rom image, ic test image , logo and default setting in this page.
natset.htm
You could set the nat setting in this page.
dmzset.htm
You could set the DMZ setting in this page.
vsset.htm
You could set virtual servers in this page.
toneset.htm
You could set tons settings in this page.
speakerset.htm
Speaker phone setting which is only for phone.
vlanset.htm
You could set the VLAN setting in this page.

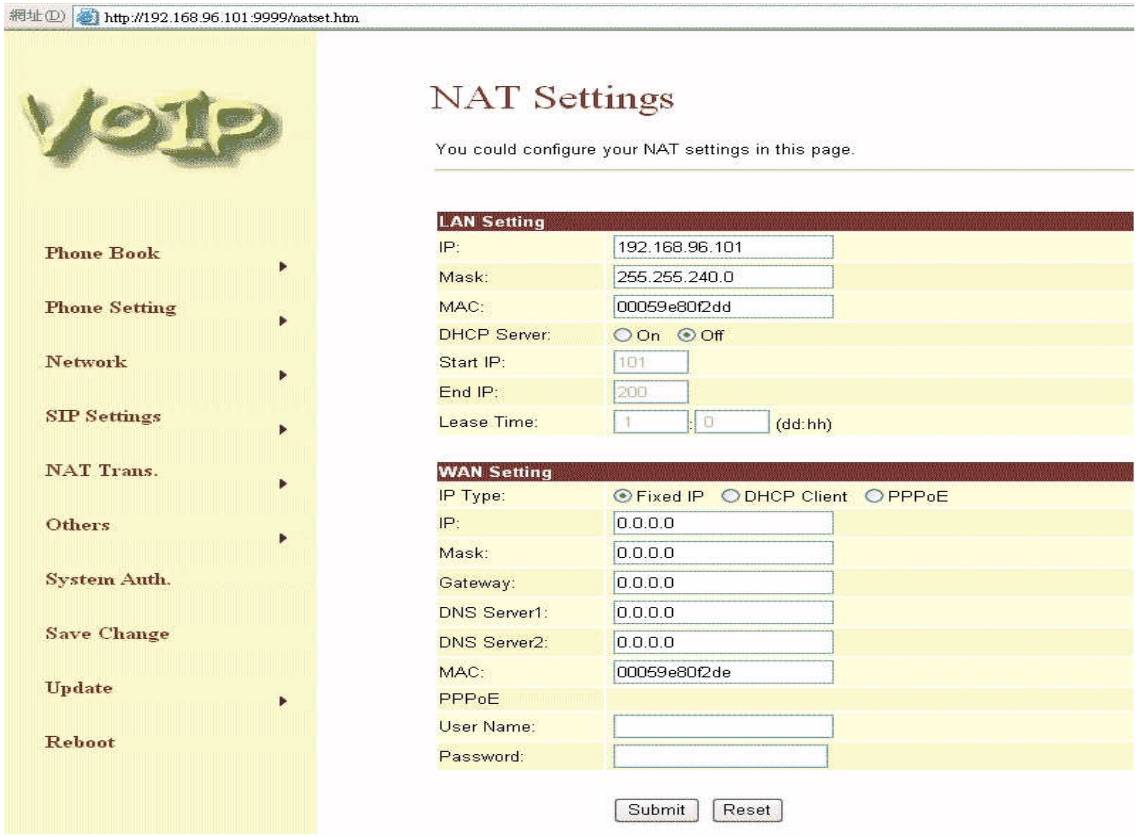
7.2 Update System

- In this page you can update the system's ROM code, IC Test , Default setting and Logo.
- Update ROM code. You can update the ROM code from this function. Please be noted that if you update the wrong file or during the update process the power is off, the system will be **crashed**.
- Update Logo. The Logo specification is "220x170" pixel and need "jepg format" file.



7.3 NAT Settings

In this page you can setup the nat function. The WAN setting is for you to set how the get the IP address for the device. The LAN setting is for the other devices to get the IP address from the device. You can choose to use DHCP server or not.

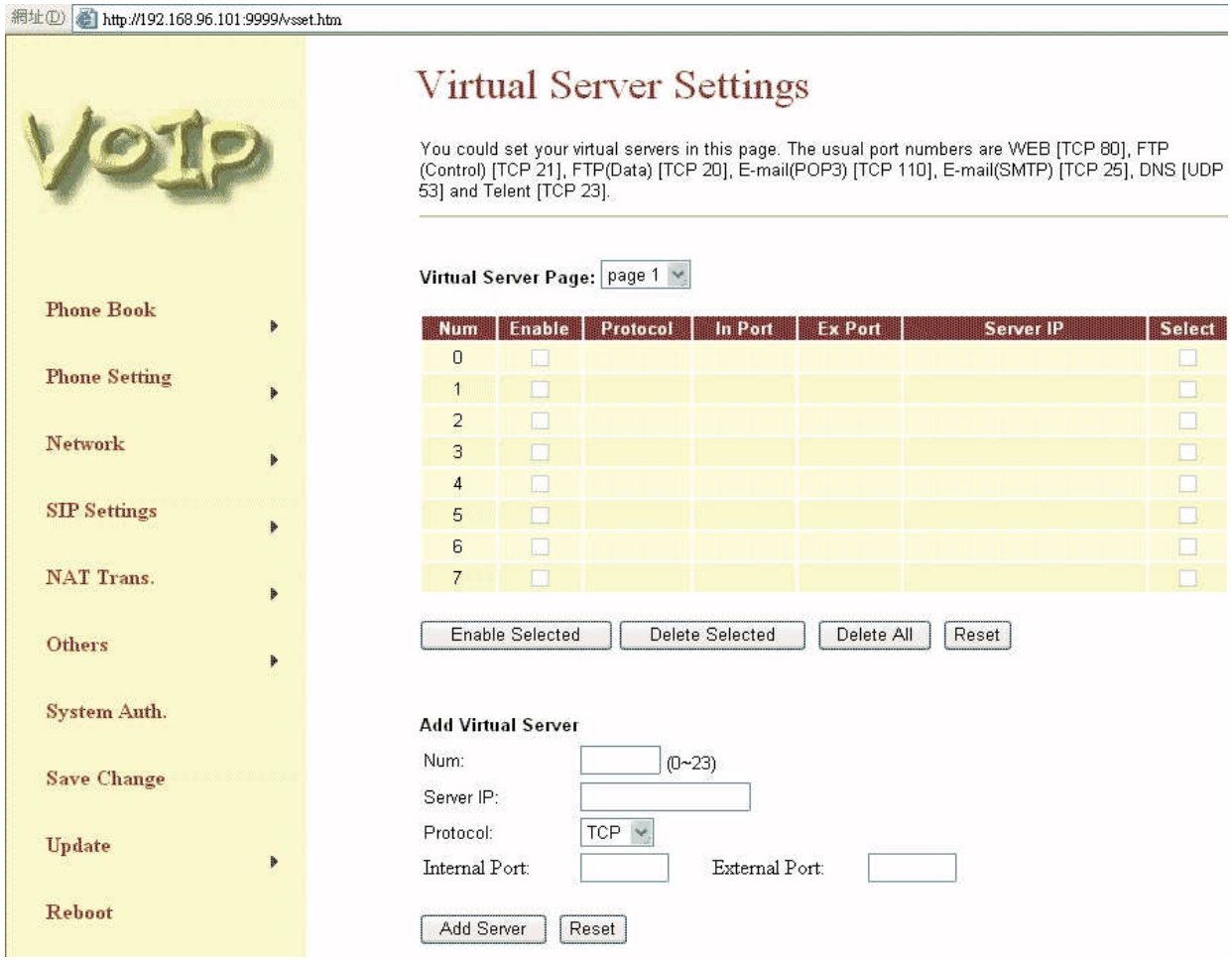


7.4 DMZ Setting

In this page you can setup the DMZ function. You need to enable/disable this function and set the IP address for DMZ.

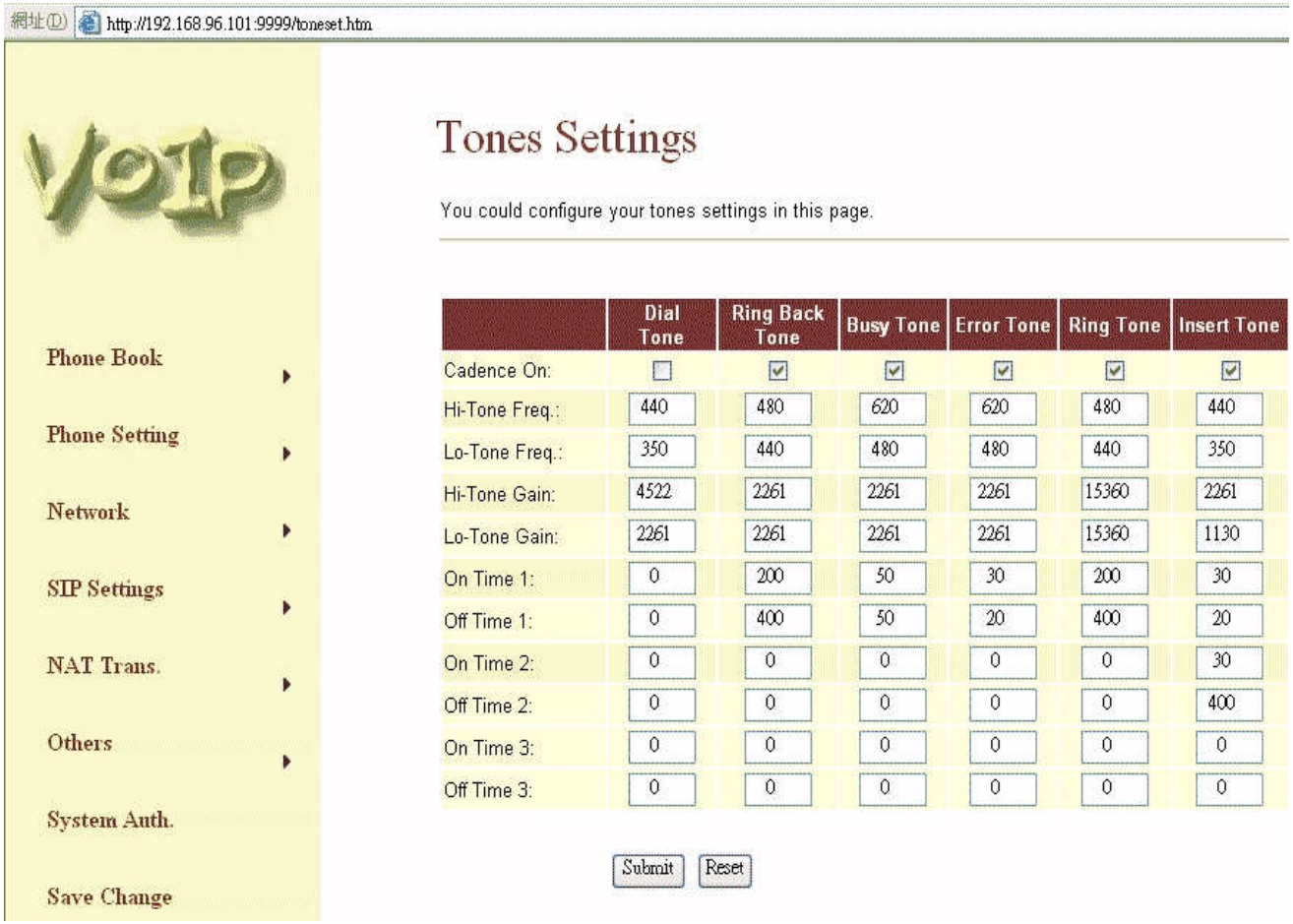
7.5 Virtual Server Settings

In this page you can setup the Virture Server function.



7.6 Tones Settings

In this page you can setup the Tone frequency and cadence to meet the requirement.

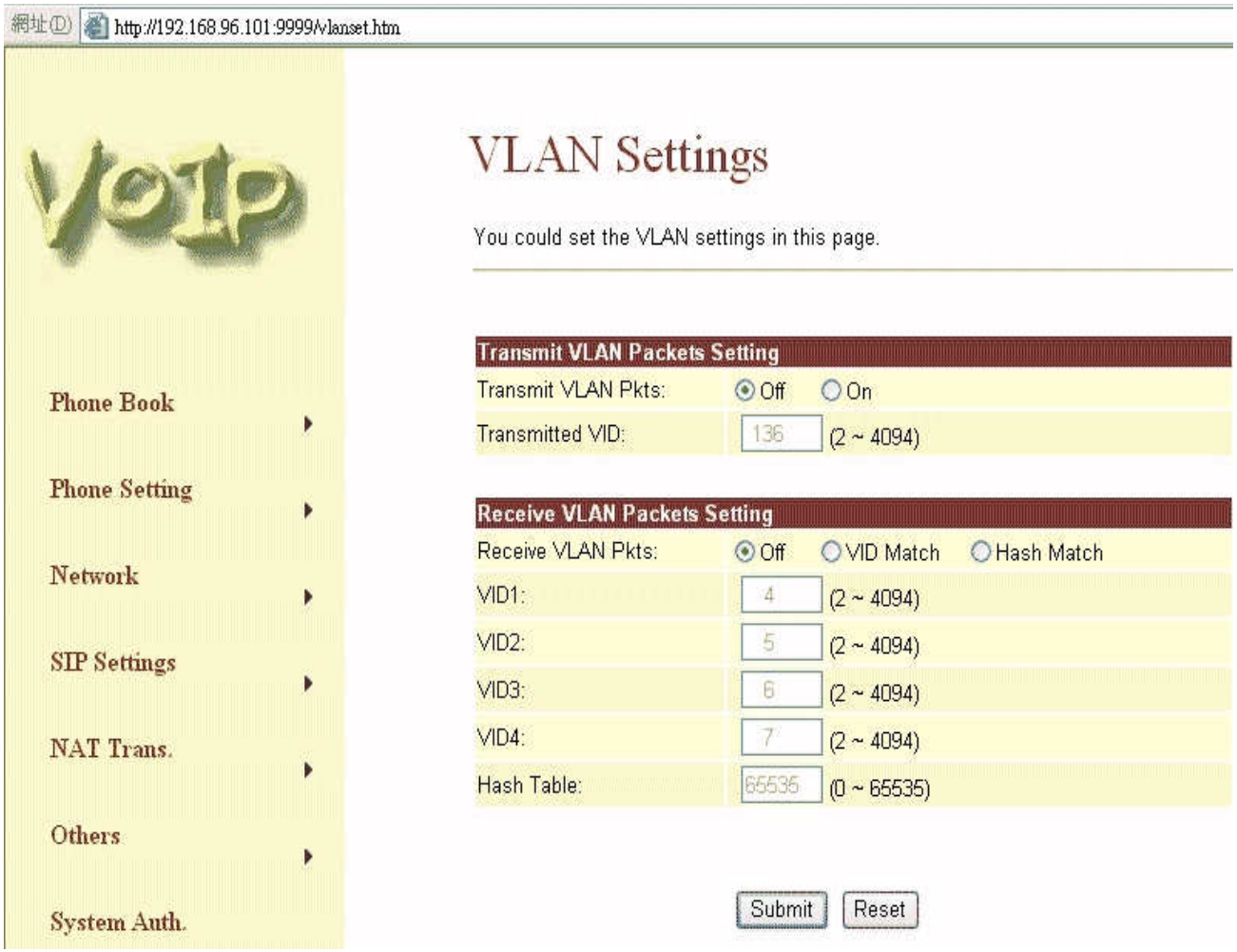

http://192.168.96.101:9999/toneset.htm
VoIP
Phone Book
Phone Setting
Network
SIP Settings
NAT Trans.
Others
System Auth.
Save Change
Tones Settings
You could configure your tones settings in this page.
Dial Tone
Ring Back Tone
Busy Tone
Error Tone
Ring Tone
Insert Tone
Cadence On:
Hi-Tone Freq.: 440 480 620 620 480 440
Lo-Tone Freq.: 350 440 480 480 440 350
Hi-Tone Gain: 4522 2261 2261 2261 15360 2261
Lo-Tone Gain: 2261 2261 2261 2261 15360 1130
On Time 1: 0 200 50 30 200 30
Off Time 1: 0 400 50 20 400 20
On Time 2: 0 0 0 0 0 30
Off Time 2: 0 0 0 0 0 400
On Time 3: 0 0 0 0 0 0
Off Time 3: 0 0 0 0 0 0
Submit
Reset

7.7 Speaker Phone Setting

In this page you can setup the Speaker function. This function only support in IP Phone with speaker phone.

7.8 VLAN Settings

In this page you can setup the VLAN function.



Chapter 8. Setup the NVP-300 series by Telnet

8.1 Login into the NVP-300 series

Please execute "Telnet + NVP-300 series IP", Then Telnet is ready to connect to the NVP-300 series. Press "Enter" and the Telnet will show the "Login: ". Input "**root**" and press the "Enter" button. Then Telnet will show the "Password: ". Input "**test**" and press the "Enter" button. Now you already login the NVP-300 series. Please follow the CLI command list to configure the NVP-300 series with proper instruction and value.

8.2 Using CLI command to configure the NVP-300 series

- **CLI command list as below:**

Item	Command	Description
1	?	Show CLI Command
2	arp	ARP Configuration
3	ipconfig	Interface Configuration
4	save	Save to flash
5	reboot	Reboot
6	exit	Exit
7	debugmode	Enter Debug Mode
8	update	Update Flash Code/RAM
9	auth	Change User Name and Password
10	nat	NAT Configuration
11	dns	DNS Configuration
12	ping	ping [-lN] [IP-addr|host-name]
13	sip	SIP Configuartion
14	ddns	DDNS Configuartion
15	sntp	SNTP Configuartion
16	vlan	VLAN Configuartion
17	time	Get System Time

Item	Command	Description
18	mactab	Show MAC Learning Table
19	dump	Read/Write Memory
20	book	Edit phone book
21	reload	Reload Factory Setting
22	watchdog	WatchDog Function
23	phone	Phone Setting
24	weblogo	Change Web's logo
25	dsp	Show dsp type
26	addport	Add Nat Port Mapping
27	cid	Select slic Cid
28	slic	read or write slic registers
29	ver	Firmware Version

Note:
"?" function is to show CLI command list in the screen.

- **ARP function list as below:**

Item	Command	Description
1	?	Show 'arp' Option
2	-a	Show ARP Table
3	-d	Delete ARP Table
4	-s	Set Static ARP Table
5	(null)	Show ARP Table

- **Ipconfig function list as below:**

Item	Command	Description
1	?	Show 'ipconfig' Option
2	-if0	Interface 0
3	-if1	Interface 1

4	-if2	Interface 2
5	-h	Set Host Name
6	-a	Set ARP Cache Expire
7	-r	Restore Current Setting
8	(null)	Show IP Setting

◆ **Ipconfig –ifN function → N is 0, 1, 2**

Item	Command	Description
1	?	Show 'ipconfig -ifN' Option
2	-t	Set Host Type
3	-m	Set MAC Address
4	-i	Set IP Address
5	-nm	Set Net Mask
6	-g	Set Gateway
7	-dns0	Set Primary DNS server
8	-dns1	Set Secondary DNS server
9	-dr	Set Default Route
10	-nat	Set NAT
11	on	Enable Interface
12	off	Disable Interface
13	-dhcps	DHCP Server Setting
14	-ddns	Set DDNS
15	-bridge	Set Bridge
16	-dev0	Set Device 0 Setting
17	-dev1	Set Device 1 Setting
18	-dev2	Set Device 2 Setting
19	(null)	Show Interface Setting

- **Save function list as below:**

Item	Command	Description
1	?	Show 'save' Option
2	-book	Save phone book
3	-sys	Save system setting

- **Reboot function is to restart the system.**
- **Exit function is to exit the CLI.**
- **Debugmode function is to enter the debugmode.**
- **update function list as below:**

Item	Command	Description
1	?	Show 'update' Option
2	-os	Update OSImage(IP filename)
3	-dsp	Update DSP Image(IP filename)
4	-all	Update All Image(IP filename)
5	-server	Update Server (IP filename length)
6	-pcm	PCM(IP filename)
7	-alaw	alaw (IP filename)
8	-ulaw	ulaw (IP filename)
9	-g729	g729 (IP filename)
10	-g723	g723 (IP filename)
11	-g726.16	g726.16 (IP filename)
12	-g726.24	g726.24 (IP filename)
13	-g726.32	g726.32 (IP filename)
14	-g726.40	g726.40 (IP filename)

Note:

IP is the TFTP server's IP address, and the filename is the image you want to download into the system.

- **Auth function list as below:**

Item	Command	Description
1	?	Show 'auth' Option
2	-admin	Change Administrator user name/password
3	-sys0	Change System user0 user name/password
4	-sys1	Change System user1 user name/password
5	-sys2	Change System user2 user name/password
6	-sys3	Change System user3 user name/password
7	-sys4	Change System user4 user name/password
8	-norm0	Change Normal user0 user name/password
9	-norm1	Change Normal user1 user name/password
10	-norm2	Change Normal user2 user name/password
11	-norm3	Change Normal user3 user name/password
12	-norm4	Change Normal user4 user name/password
13	-ppp	Change PPP user name/password
14	(null)	Show auth Setting

In each item includes

Item	Command	Description
1	?	Show 'auth' Option
2	-user	Change User Name.'auth -sys3 -user xxx '
3	-pass	Change Password. 'auth -sys3 -pass xxx xxx'
4	(null)	Show auth's System/PPP Setting

Note:

If you want to change the password, you need to type the password twice in the CLI.

- **Nat function list as below:**

Item	Command	Description
1	?	Show 'nat' Option
2	-vs	Set 'nat -vs' Option
3	-dmz	Set 'nat -dmz' Option
4	(null)	Show NAT Setting

In DMZ item includes

Item	Command	Description
1	?	Show 'nat -dmz' Option
2	on	EnableDMZ
3	off	EnableDMZ
4	-ip	Set DMZ IP address
5	(null)	Show DMZ Setting

- **Dns function list as below:**

Item	Command	Description
1	?	Show 'dns' Option
2	-q	DNS query. dns -q domain-name
3	(null)	Show DNS Table

- **Ping function list as below:**

Item	Command	Description
1	?	Show 'ping' Option
2	-l	ping [-l N] [IP-addr|host-name]
3	(null)	ping [IP-addr|host-name]

- **Sip function list as below:**

Item	Command	Description
1	?	Show 'sip' Option
2	-proxy0	sip -proxy0

3	-proxy1	sip -proxy1
4	-proxy2	sip -proxy2
5	-upnp	sip -upnp on/off/show
6	-exts	sip -exts sip upnp external-port
7	-extr	sip -extr rtp upnp external-port
8	-sipp	sip udp port
9	-rtpp	sip rtp port
10	-stun	sip -stun on/off
11	-rport	sip -rport on/off
12	-sserver	sip -sserver stun-server
13	-out	sip -out outbound-proxy
14	-dump	sip –dump
15	-log	sip -log on/off
16	-drtp	sip -drtp 0/1/2
17	-rtpnc	sip -rtpnc on/off
18	-wanip	sip –wanip
19	-nattype	sip –nattype
20	-hbyrfc	sip –hbyrfc
21	-dereg	sip -dereg
22	-restart	sip -restart
23	-jbt	sip -jitter buffer Threshold
24	(null)	Show SIP Setting

- **Ddns function list as below:**

Item	Command	Description
1	?	Show 'ddns' Option
2	-type	Set DDNS Type
3	-host	Set Host Name
4	-wild	Set Wild Card Mode
5	-mx	Set Mail Exchanger

6	-backmx	Set Mail Exchanger Mode
7	-offline	Set Offline Mode
8	-user	Set Login User Name
9	-pass	Set Login Password
10	(null)	Show DDNS Setting

- **Sntp function list as below:**

Item	Command	Description
1	?	Show 'sntp' Option
2	-on	Enable SNTP Client
3	-off	Disable SNTP Client
4	-ip1	Set SNTP Server1 IP
5	-ip2	Set SNTP Server2 IP
6	-mode	Set SNTP Client Mode
7	-zone	Set GMT Time Zone: [+|-][hour]:[min]
8	-adjust	Set Adjustment Time: [second]
9	(null)	Show SNTP Setting

- **Vlan function list as below:**

Item	Command	Description
1	?	Show 'vlan' Option
2	-tx	Tx Vlan setting
3	-rx	Rx Vlan setting
4	(null)	Show Vlan Setting

- **Time function list as below:**

Item	Command	Description
1	?	Show 'Time' Option
2	-t	Modify Time: hour:min:sec
3	-d	Modify date: year:mon:date

4	(null)	Show Data & Time

- **Mactab function is to show MAC learning table.**

- **Dump function list as below:**

Item	Command	Description
1	?	Show 'dump' Option
2	-r	dump -r XXXXxxxx
3	-w	dump -w XXXXxxxx XX

- **Book function list as below:**

Item	Command	Description
1	?	Show 'book' Option
2	-a	Show answer list
3	-c	Show call list
4	-s	speed dial
5	-p	phone book

- **Reload function is to Reload Factory Setting, please make sure you want to do the factory reset.**

- **Watchdog function list as below:**

Item	Command	Description
1	?	Show 'WatchDog' Option
2	on	Enable WatchDog
3	off	Disable WatchDog
4	(null)	Show WatchDog Setting

- **Phone function list as below:**

Item	Command	Description
1	?	Show 'phone' Option

2	-autoanswer	phone auto answer
3	-vol	Volume setting
4	-block	Block Incoming call
5	-ring	Set Melody Ringer
6	-forward	Auto-forward Incall to Phone[0-9] in Book
7	(null)	Show Phone Setting

- **Weblogo function list as below:**

Item	Command	Description
1	?	Show 'weblogo' Option
2	-on	Vender Logo
3	-off	Crystal media Logo
4	(null)	Show weblogo Setting

- **Dsp function is to show dsp code type.**

- **Dport function is to add Nat Port Mapping 8.3.1.27 cid function**

Item	Command	Description
1	?	Show 'cid' Option
2	-off	Disable Slic Cid signal
3	-1	Tx FSK after 1st Ring
4	-2	Tx FSK before 1st Ring
5	-3	Tx DTMF before 1st Ring
6	-4	Tx FSK with Line reversal before 1st Ring
7	-5	Tx DTMF with Line reversal before 1st Ring

8	-time	FSK cid with time mseeage
9	-single	Single type FSK CID
10	(null)	Show Cid Option

- **Slic function list as below:**

Item	Command	Description
1	?	Show 'slic' Option
2	-ring	Issue Ring signal
3	-r	read slic addr
4	-w	write slic addr
5	-a	read all slic reg
6	(null)	Show slic register

- **Ver function is to show Firmware Version.**

Chapter 9. How to make a phone call

When your NVP-300 series is configured properly, you can make a phone call to your friend in the same Service provider.

If you want to make a phone call, you can dial the phone number and press "#" button to start to dial the phone number.

The NVP-300 also provides some functions that list as below:

1. **Call Waiting:** When a new call is coming while you are talking, you can push the Flash button to switch to the new call. You can push the Flash button to switch between the two calls.
2. **Call Hold:** You can push the Hold key to hold the current call for a while, then push Hold key again to keep talking.
3. **3-way conference:** If you want to make a 3-way conference call, you can make a phone call to the first phone number. After the call is established, push the Flash button then you can hear the Dial tone, then make a phone call to the second phone number. When the second call is established, press the Flash button again. In the 2-port NVP-300, only one FXS port can support 3-way conference function, and one is G.723/G.729A, the other one is G.711.

Chapter 10. Get a FWD account

1. The website is www.freeworlddialup.com; you can apply an account to use the VoIP communication. You can follow the instruction to input the information. After you finished, you will receive a mail sent by the FWD mail system, you will get the account information in the mail.

2. When you got the account, you can setup the related information into the NVP-300 series.

3. You can setup the related information into the NVP-300 series by web browser. Also you can use Telnet, Console via CLI command to configure the NVP-300 series. You need to input the Proxy Name, Domain Name, Register Name, and password. The Display Name you can input what you want to let others see.

4. After you registered to the SIP Server, you can try to call your friends who also registered in the same SIP Server. You just need to dial your friend's user name (registered name) and press "#" then you can make a phone call to your friend.

5. If you want to make a phone call to the other in the internet, first you need to registered in a Proxy Server (with SIP Server IP, Domain IP, registered name, Password), make sure you already enable Stun function, then you can try.

Chapter 11. How to change from FXS to FXO

Please pick up hand set, then dial "0" + "＊" key when you would like to change from FXS to FXO.
Please press hook switch once when you would like to return FXS.

Note:
NVP-300SO/SSO default on FXS mode.

Appendix A: Dial Plans

The SIP code will allow provisioning (via web browser) of the dial plan. A dial plan gives the unit a map to determine when a complete number has been entered and should be passed to the gatekeeper for resolution into an IP address. Dial plans are expressed using the same syntax as used by MGCP NCS specification.

The formal syntax of the dial plan is described by the following notation:
Digit ::= "0" | "1" | "2" | "3" | "4" | "5" | "6" | "7" | "8" | "9"
Timer ::= "T" | "t"
Letter ::= Digit | Timer | "#" | "*" | "A" | "a" | "B" | "b" | "C" | "c"| "D" | "d"
Range ::= "X" | "x" -- matches any digit
| "[" Letters "]" -- matches any of the specified letters
Letters::= Sub range | Sub range Letters
Sub range::= Letter -- matches the specified letter
| Digit "-" Digit -- matches any digit between first and last
Position::= Letter | Range
String Element::= Position -- matches any occurrence of the position
| Position "." -- matches an arbitrary number of occurrences
including 0
String ::= String Element | String Element String
String List::= String | String "|" String List
Dial Plan::= String | "(" String List ")"
A dial plan, according to this syntax, is defined either by a (case insensitive) string or by a list of strings. Regardless of the above syntax a timer is only allowed if it appears in the last position in a string (12T3 is not valid). Each string is an alternate numbering scheme. The unit will process the dial plan by comparing the current dial string against the dial plan, if the result is under qualified (partial matches at least one entry) then it will do nothing further. If the result matches or is over-qualified (no further digits could possibly produce a match) then send the string to the gatekeeper and clear the dial string. The Timer T is activated when it is all that is required to produce a match. The period of timer T is 4 seconds. For example a dial plan of (xxxT|xxxxx) will match immediately

if 5 digits are entered, it will also match after a 4-second pause when 3 digits are entered.

Simple Dial Plan

Allows dialing of 7 digit numbers (e.g. 5551234) or an operator on 0. Dial plan is (0T|xxxxxxx)

Non-dialed Line Dial Plan

As soon as handset is lifted, the unit contacts the gatekeeper (used for systems where DTMF detection is done in-call). Dial plan is (x.) i.e. match against 0 (or more) digits. Note: the dot '**.**'

Complex Dial Plan

Local operator on 0, long distance operator on 00, four digit local extension number starting with 3,4 or 5, seven digit local numbers are prefixed by an 8, two digit star services (e.g. 69), ten digit long distance prefixed by 91, and international numbers starting with 9011+variable number of digits.

Dial plan for this is:

(0T|00T|[3-5]xxx|8xxxxxxx|*xx|91xxxxxxxxxx|9011x.T)

Replace prefix code:	On Off
Replace rule:	001+009+006 -> 005
Dial Plan:	*xx+#xx+10x+11x+xxxxxxxx
Auto Prefix:	02 (0000~9999)
Prefix Unset Plan:	1+0+xxxx+xxxxxx
Auto Dial Time:	5 (3~9 sec)

Symbol explan:

x or X	0,1,2,3,4,5,6,7,8,9
+	or

Replace rule: If replace prefix code is ON and prefix number is matched with rule then 005 will replace prefix.

Auto Dial Time : Stop dialing after seconds then send dial number out

Dial Plan: When match with pattern then send dial number out but if fisrt digit is '0' then dial plan will be ignored.

Example:

*xx	If matched with one of *00,*01….*99 then will send number out
#xx	If matched with one of #00,#01….#99 then will send number out
10x	If matched with one of 100,101….109 then will send number out
11x	If matched with one of 110,111….119 then will send number out
Xxxxxxxx	If dial with 8 digits then send number out

Auto Prefix : Number for add before dial number

Prefix Unset Plan : When first digit or dial number match with pattern then ignore auto prefix

0	Iignore auto prefix if first digit is '0'
1	Ignore auto prefix if first digit is '1'
Xxxxx	dial numbers are 4 digits ignore auto prefix
Xxxxxx	dial numbers are 5 digits ignore auto prefix

Appendix B: Cable Requirement

A CAT 3,4 or 5 UTP (unshielded twisted pair) cable is typically used

To connect the Ethernet device to the modem.

A 10Base-T cable often consists of four pairs of wires, two of which are used for transmission. The connector at the end of the 10Base-T cable is referred to as an RJ-45 connector and it consists of eight pins.

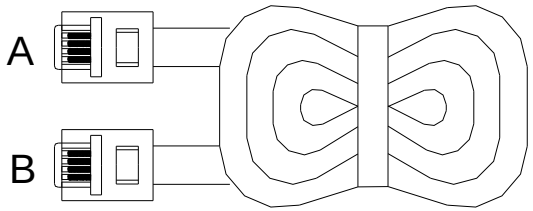
The Ethernet standard uses pins 1,2,3 and 6 for data transmission purposes.

Table RJ-45 Ethernet Connector Pin out Assignments

PIN	MNEMONIC	FUNCTION
1	TX+	Ethernet differential Transmit signal(+)
2	TX-	Ethernet differential Transmit signal(-)
3	RX+	Ethernet differential receive signal(+)
4	NC	Unused
5	NC	Unused
6	RX-	Ethernet differential receive signal(-)
7	NC	Unused
8	NC	Unused

Standard telephone wire of any gauge or type-flat, twisted or quad is used to connect the Modem to the telephone network. A telephone cable typically consists of three pairs of wires, one of which is used for transmission. The connector at the end of the telephone cable is called RJ-11 connector and it consists of six pins. POTS (plain old telephone services) use pins 3 and 4 for voice transmission. A telephone cable is shown below.

Figure Telephone cable



The A and B connectors on the rear of the modem are RJ-11 connectors. These connectors are wired identically. The RJ-11 connectors have six positions, two of which are wiring, The Modem uses the center two pins. The pin out assignment for these connectors is presented below.

Table RJ-11 Pin out Assignments

Pin#	MNEMONIC	FUNCTION
1	NC	Unused
2	NC	Unused
3	TIP	POTS
4	RING	POTS
5	NC	Unused
6	NC	Unused

Appendix C: Product Specification

Product Name：VOIP with Router Adaptor
Application: Home networking solution

Product Specification：

Hardware

- Digital Signal Processors & Control Processor System On Chip (SOC) for Network Processing and DSP Application
  MIPS-X5 unified RISC and DSP core (up to 180 DSP MIPS)
  384K bytes on-chip RAM, 16-way interleaved with single cycle access
  16-K byte cache
  Low power, 1.8V core voltage, 3.3V I/O voltage
  2M bytes flash memory

- I/O
- 2 Standard 10/100 Base-TX RJ 45 interface for 2 FXS model
  2 RJ 11 Loop Start interfaces for FXS

- Mechanical, Environment & Power
  Dimension: L x W x H =184mm x 146mm x 40mm
  Operating temperature: 0℃ to 50℃ (32 to 122 F)
  Operating humidity: 10% to 95% (non-condensing)
  Storage temperature: -10 to 60℃ (14 to 140F)

AC-to-DC power supply (12VDC, 120 VAC, 60 Hz. For US or 12VDC, 230VAC,50Hz for Europe)
Power consumption : 5.5 watt (Typical)

- Compliant
  CE
  FCC part 15 A

Software

- Compression algorithms: ITU G. 711, G.723, G726, G.729A/B and T.38 Fax Relay
  Hybrid echo cancellation G.168 (16 ms)
  Auto switch between Fax and voice
  DTMF tone detection/regeneration
  Comfort Noise Generation (CNG)
  User programmable Call Progress detection/generation
  Voice Activity Detection (VAD)

- Fax
  Facsimile protocol(option): T.38
  Real-time fax over IP
  DTMF tone detection/regeneration

- Management Tools
  HTTP Server
  TFTP and flash memory for remote software download and upgrade

- SIP Protocol Stack

  Compliant with SIP v2.0 (RFC 3261)

- MGCP Protocol Stack
  Compliant with MGCP protocol specifications (RFC 3435)

- Lan / Wan Functions
  Tagging VLAN(IEEE802.1q)
  QOS (IEEE802.1p)
  DHCP Server
  PPPOE
  NAT
  Firewall
  SNMP(V1/V2)

- Chipset:Crystal Media

Appendix D: Troubleshooting

The NVP-300 can be easily monitored through its comprehensive panel indicators. These indicators assist the network manager in identifying problems the hub may encounter. This section describes common problems you may encounter and possible solutions

1. Symptom:	Power LED indicator does not light up (green) after power on.
Cause:	Defective External power supply
Solution:	Cheek the power plug by plugging in another that is functioning properly. Check the power cord with another device. If these measures fail to resolve the problem, have the unit power supply replaced by a qualified distributor.

2. Symptom:	WAN/LAN LED indicator does not light up (green) after making a connection.
Cause:	Network interface (e.g, a network adapter card on the attached device), network cable, or switch port is defective.
Solution:	2.1 Power off for the NVP-300. 2.2 Verify that the switch and attached device are powered on. 2.3 Be sure the cable is plugged into both the switch and corresponding device. 2.4 Verify that the proper cable type is used and its length does not exceed specified limits. 2.5 Check the NVP-300 on the attached device and cable connections for possible defects. 2.6 Replace the defective NVP-300 or cable if necessary.

3. Symptom:	Phone 1/2 LED indicator does not light up (green) after making a connection.
Solution:	3.1 Be sure the phone wire is plugged into both the NVP-300 and phone set. 3.2 sure the phone set is analog type and on hang on status 3.3 ck the NVP-300 on the phone set and cable connections for possible defects 3.4 Replace the defective NVP-300 or phone set if necessary

System Diagnostics

Power and Cooling Problems

If the POWER indicator does not turn on when the power cord is plugged in, you may have a problem with the power outlet, power cord, or internal power supply as explained in the previous section. However, if the unit power is off after running for a while, check for loose power connections, power lost or surges at the power outlet, and verify that the fan on back of the unit is unobstructed and running prior to shutdown. If you still cannot isolate the problem, then the internal power supply may be defective. In this case, contact your dealer.

Installation

Verify that all system components have been properly installed. If one or more components appear to be malfunctioning (e.g., the power cord or network cabling), test them in an alternate environment where you are sure that all the other components are functioning properly.

Transmission Mode

The default method of selecting the transmission mode for RJ-45 ports is 10/100 Mbps ETHERNET, for RJ-11 port are Voice, Therefore, if the Link signal is disrupted (e.g., by unplugging the network cable and plugging it back in again, or by resetting the power), the port will try to reestablish communications with the attached device via auto-negotiation. If auto-negotiation fails, then communications are set to half duplex by default. Based on this type of industry-standard connection policy, if you are using a full-duplex device that does not support auto-negotiation, communications can be easily lost (i.e., reset to the

wrong mode) whenever the attached device is reset or experiences a power fluctuation. The best way to resolve this problem is to upgrade these devices to a version that supports Ethernet and analog phone set.

Physical Configuration

If problems occur after altering the network configuration, restore the original connections, and try to track the problem down by implementing the new changes, one step at a time. Ensure that distance of cable and other physical aspects of the installation do not exceed recommendations

System Integrity

As a last resort verify the switch integrity with a power-on reset. Turn the power off and then on several times. If the problem still persists after you have completed all the preceding diagnoses, contact your dealer

FAQ:

1 **Problem:**	This is the first time for me to configure VOIP gateway, unfortunately your product manual relate with persons who had some previous knowledge about VOIP protocols, SIP and others. So all I want to learn now is how can I connect two phones connected to two gateways on LAN, and how can I do the same on WAN For example: if I will connect two gateways on the same LAN and I will give the first IP 192.168.1.1 and the other 192.168.1.2, and I will connect one phone set to each gateway, so how can I communicate between the two phone sets. How can I dial?
Answer:	First,the checks your device whether without any error .Then get in your computer TCP/IP setting,you must choose"Obtain WAN configuration dynamically".Finally,reference our VoIP_User_Manual A.3(WAN,LAN,and SIP setting) to configuration dynamically.

2 **Problem:**	Attached my account details with a VOIP service provider Please tell me what settings I have to enter in the SIP configuration page in all fields especially in ◆ RTP port number ◆ Dial Plan ◆ Line1 Gain Control ◆ Tone (Case after making any configuration I have a very strange tone when I pick up the handset, it seems like busy tone but its faster)
Answer:	First,you must register account number on SIP server. Then you'll get the user name,password..,immediately after open the browser to inputing "192.168.1.1".(But the premise is your computer necessarily must choose "Obtain WAN configuration dynamically" on TCP/IP setting.) After inputing the user number and password,you'll configure the setting for this issue.You can set the user1&user2's phone number in the SIP configuration page.
Note:	We suggest you don't need to setting it.(RTP port number,Dail plan,Line1 Gain Control,Tone)

3 **Problem:**	I have already registered in SIP VOIP service, and I have send to you my account details including number, User name, password and SIP settings. Also I have obtained the WAN to be configured dynamically, and it obtained an IP I have configured the server settings in the SIP configuration page, and after configuring the server settings, the tone of the phone changes, it gives me a tone as the busy tone (but its faster). Also I have set the User1 Phone Number, including User name & Password (Attached) Also I didn't setting any of (RTP port number, Dial plan, Line1 Gain Control, Tone)
Answer:	It needs to open ˇ (Send Registration Request with Expire Time) function to select to register sip server. This function lies in sip server page.

4 **Problem:**	Could you tell me whether NVP-300SO supports Caller ID for PSTN? Our customer tests NVP-300SO and he tells that NVP-300SO supports Caller ID for IP Network, but it doesn't support Caller ID for PSTN (he uses FSK). He tells that NVP-300SO doesn't send Caller ID to IP. Could you check it? Why isn't there any informations about NVP-300SO and NVP-300SSO on you webside?
Answer:	You can set NVP-300SO to show Caller ID in your PSTN Phone or IP Phone, we support four of Caller ID mode, such as no display call ID; display call ID before ring; display call ID after ring; and support DTMF modulation. It must be base on your environment to set properly the Caller ID mode for FSK or DTMF, as your customer set that on FSK was failure, please try to set that on DTMF.

5 **Problem:**	This is the first time for me to configure VOIP gateway, that have some noise when I want to dial. How can I dial?
Answer:	There is this kind of phenomenon is normally, please press "redial" when you to meet this kind of condition.

6 **Problem:**	If the DHCP using default set in the device? Mean if I first time connect device and have DHCP in the network, he must get the ip? Is the dial tone start only when network settings ok?
Answer:	If you just would like to got Dial tone start, you don't need to change anything, you just prepare one analog phone set connectity to phone 1 or phone2 of NVP-300SSO, now you get busy tone, if no busy tone please check between phone set and NVP-300SSO both connecting whether ok. Then you must prepare one Ethernet link to LAN or WAN port of NVP-300SSO, and LAN/WAN port link LED should be on. If you don't have to connect LAN or WAN, you can't got dial tone start from FXS port by phone1 or phone2.

7 **Problem:**	Our costumer wants to have 2 VLAN's on 1 (WAN) port. He wants 1 VLAN for voice and 1 VLAN for data. Is it possible to configure to configure 2 VLAN's on 1 phisical WAN port? And how to do it? Olny in VLAN Configuration section? or maybe also in WAN Configuration?
Answer:	NVP-300 series only provide one LAN port, so it can't be setted VLAN groups.

Appendix E: Compliance and Safety Information

FCC Radio Frequency Interference Statement

This equipment has been tested and found to comply with the limits for a computing device, pursuant to Part 15 of FCC class B rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generate, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

1. Reorient or relocate the receiving antenna.
2. Increase the separation between the equipment and receiver.
3. The equipment and the receiver should be connected to outlets on separate circuits.
4. Consult the dealer or an experienced radio/television technician for help.

Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

If this telephone equipment causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice isn't practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.
The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the proper functioning of your equipment. If they do, you will be notified in advance in order for you to make necessary modifications to maintain uninterrupted service.

This equipment may not be used on coin service provided by the telephone company. Connection to party lines is subject to state tariffs.

Important Safety Instructions

- **Caution:** The direct plug-in wall transformer serves as the main product for disconnecting. The socket outlet shall be installed near the product and be readily accessible.
- **Caution:** Use only the power supply included with this product. In the event the power supply is lost or damaged:In the United States, use only with CSA certified or UL listed Class 2 power supply.
  IN Europe, use only with CE certified power supply.
- **Do not** use this equipment near water, for example in a wet basement.
- **Avoid** using a telephone during an electrical storm. There may be a remote risk of electrical shock from lightning.
- **Do not** use the telephone to report a gas leak in the vicinity of the leaking area.
- If you experience trouble with this unit, please contact customer service of your dealer immediately.
- **DO NOT DISASSEMBLE THIS EQUIPMENT**. It does not contain any user serviceable components.

FCC Warning

This equipment has been tested to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment can generate, use, and radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at owner's expense.

CE Mark Warning

This is a class B product. In a domestic environment, this product may cause radio interference in which case the user may be required to take adequate measures.

WEEE Warning

To avoid the potential effects on the environment and human health as a result of the presence of hazardous substances in electrical and electronic equipment, end users of electrical and electronic equipment should understand the meaning of the crossed-out wheeled bin symbol. Do not dispose of WEEE as unsorted municipal waste and have to collect such WEEE separately.

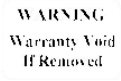
Warranty

The original owner of this package will be free from defects in material and workmanship for one-year parts after purchase. For the warranty to apply, you must register your purchase by returning the registration card indicating the date of purchase.

There will be a minimal charge to replace consumable components, such as fuses, power transformers, and mechanical cooling devices. The warranty will not apply to any products which have been subjected to any misuse, neglect or accidental damage, or which contain defects which are in any way attributable to improper installation or to alteration or repairs made or performed by any person not under control of the original owner.

The above warranty is in lieu of any other warranty, whether express, implied, or statutory, including but not limited to any warranty of merchantability, fitness for a particular purpose, or any warranty arising out of any proposal, specification, or sample. Shall not be liable for incidental or consequential damages. We neither assumes nor authorizes any person to assume for it any other liability.

Note: Please do not tear off or remove the warranty sticker as shown, otherwise the warranty will be void.



Chinese SJ/T 11364-2006

部件名称	有毒有害物质或元素					
	铅(Pb)	汞(Hg)	镉(Cd)	六价铬[Cr(VI)]	多溴联苯(PBB)	多溴二苯醚(PBDE)
结构壳体	○	○	○	○	○	○
电路组	○	○	○	○	○	○
电源供应器	○	○	○	○	○	○
风扇	○	○	○	○	○	○
线材	○	○	○	○	○	○
包装及配件	○	○	○	○	○	○

○：表示该有毒物质在该部件所有均质材料中的含量均在 SJ/T 11364/2006 标准规定的限量要求以下。
╳：表示该有毒物质至少在该部件的某依均质材料中的含量超出 SJ/T 11364-2006 标准规定的限量要求。

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